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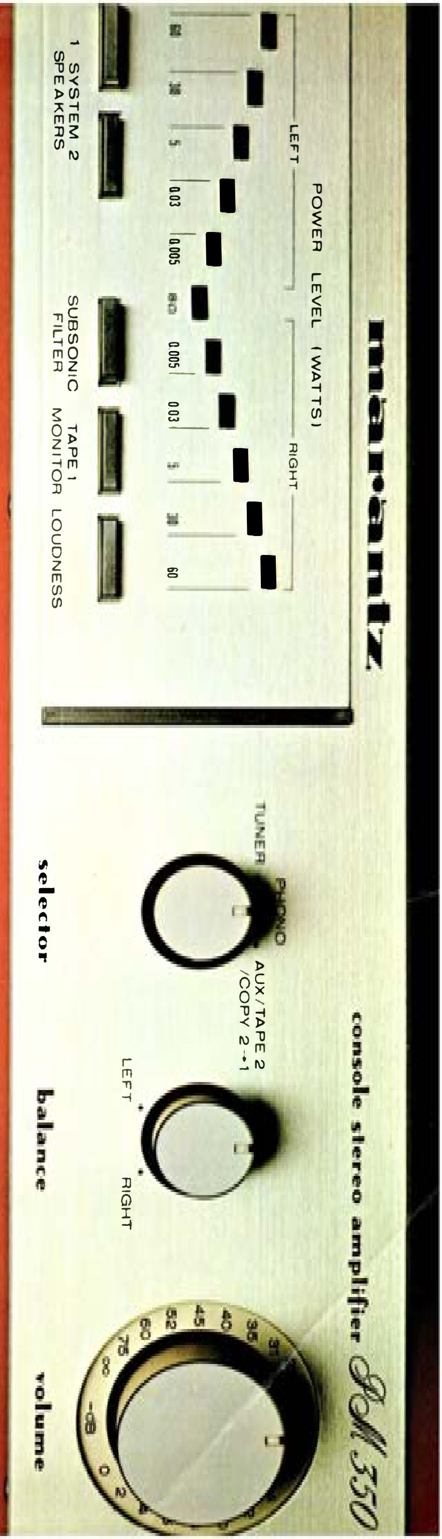
hi-fi buyer's guide

AMPLIFIERS

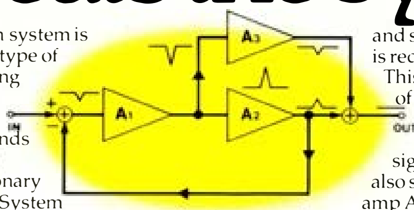


FULL TEST REPORTS ON 50 AMPLIFIERS

PLUS MOVING COIL HEAD AMPLIFIERS AND TRANSFORMERS

BY STAN CURTIS

The System that beats the system.



Arguments over which system is best (or worst) for which type of distortion have been raging for years.

But now, Sansui have perfected a system that ends disputes once and for all.

Because their revolutionary new Super Feedforward System is designed to eliminate *all* kinds of distortion. At *all* frequencies.

What Super FF does

Distortions such as harmonic, intermodulation, cross over, switching, TIM and envelope have been virtually eliminated.

And even the, as yet, unknown and unquantifiable types of distortion are guarded against. (TIM was once considered one of these.)

How Super FF works

The diagram shows how a distortion, generated in A2, is returned to the input in reverse phase, where it is added to A1. The reverse phase signal is then amplified by A1 and sent to A2. Thus distortion is reduced at the output of A2. This is the working principle of NFB.

In the Super Feedforward system, a reverse-phase signal at the output of A1 is also sent to error correction amp A3, where it is amplified and then sent on to the output (rather than the input) of A2.

In this way the feedforward circuit removes what little distortion NFB fails to eliminate.

The Super FF Unit

For some time Sansui have been considered the pacesetters in advanced audio technology.

The development of the Super Feedforward system confirms this.

But the new AU-D9 Super FF amp is not only a remarkable improvement that makes your music sound better – it is also remarkable value for money.

Write for details and we'll tell you about our matching tuners as well.

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The end of distortion disputes

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AUG 1982 No 4 IN A SERIES BY

HI FI FOR PLEASURE

hi-fi buyer's guide

TO AMPLIFIERS By STAN CURTIS

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INTRODUCTION

The task of writing this Buyer's Guide to amplifiers proved to be far more demanding of time and stamina than I ever imagined. For nearly three months I lived, ate, and drank amplifiers and even, on occasion, fell asleep amongst them. Each model was tested in the laboratory, opened up and examined, and then dropped into my hi-fi system for a listening audition; the time allocated to the latter sometimes depending on how pleasant, or not, I found the experience!

In producing the Guide I've tried to avoid the temptation to compile a short-list of these few nice-sounding models which gain regular rave reviews. In practice people buy amplifiers for any number of different reasons. They may want a model whose styling is attractive; they may do a lot of tape-recording and so seek comprehensive dubbing facilities; they may give a lot of parties and need a high-power model which can drive several pairs of loudspeakers; or they might just want an amplifier offering excellent sound quality — all other considerations being irrelevant.

The familiar Star Rating system has been borrowed from my *Hi Fi for Pleasure* reviews but it is important that readers do not rely solely on these and ignore the detailed comment. Otherwise there would be a temptation to turn to the 5-Star models and ignore the rest. It must be remembered that the stars only rate an amplifier's 'merit' in terms of overall performance, both measured and subjective; while the 'value' rating considers what a particular product offers in relation to its competitors. The fine print will be of more help in outlining the facilities offered; the standard of construction and with it a guide to the probable reliability; and some consideration of possible compatibility problems in the choice of ancillary equipment.

Some readers will be aware that over the last 15 years or so I have designed quite a few hi-fi products and, indeed, I still do a fair amount of design work through my consultancy company. However, I do have a policy of keeping myself the proverbial barge-pole away from products with which I've had some involvement, which means that a few models have not appeared in these pages. Contrary to belief, this sort of consultancy work enables me to give a far better service to readers, as I'm able to keep well abreast of new techniques; gain first hand experience of why some design approaches succeed and others fail; not least of all, maintain a test-laboratory equipped with some £50,000 worth of modern instruments capable of putting the manufacturer's claim to the test.

Finally my thanks go to Dennis and my Hewlett-Packard computer, who between them did most of the laboratory testing; Barbara, who typed the best part of 80,000 words; and Angela, my wife, who went without any social life for the months spent listening until the early hours. If you find the Guide useful and helpful, it will all have been worthwhile.

USING THIS BUYER'S GUIDE

The review of each amplifier contains a lot of useful information broken down into a number of sections. The introductory paragraphs are mainly descriptive, with an impression given of the product's styling and standard of finish, plus a list of the main facilities and special features offered by the manufacturer. My initial intentions of producing a master index listing all the facilities and features were abandoned when it was realised just how complex and unreadable this would have to be to cover all the models.

The second section covers 'Construction & Technology'. We open up the amplifier and study the constructional standards, the quality of the components used, and such oft-forgotten aspects as the safety of the mains supply wiring and connections. Such an inspection is, unfortunately, the only guide we can give to potential unreliability, for the majority of amplifiers sold are only on sale for 12 to 15 months, and by the time we have incontrovertible evidence of unreliability, the model is being phased out. What we can do, though, is to point out which models are badly constructed and therefore more liable to develop faults.

Under 'Technology' we look for any novel or interesting circuit techniques used in the design of the amplifier. Of necessity these descriptions must be short and so much simplified, but if the technique is considered interesting enough it will have been written-up in detail as part of the 'Short Circuit' series of articles in *Hi-Fi for Pleasure*.

We next look at the measured results of bench testing in the laboratory. These results are, for the large part, presented as a series of bar graphs giving 'at a glance' comparative performance as well as actual values derived by using the printed scales. These bars are supplemented by three graphs and some additional written data, and are best understood by making a thorough reading of the 'Measurements' introductory article. As a general guide to format all the bars are arranged so that higher is better.

The laboratory results are summarised and discussed in the written text with again emphasis on those figures which are particularly good or, more likely, particularly poor. Again, as a generalisation, if the result for a particular test is not discussed it means that it was adequate and not worthy of further comment. The written summary should enable you to get an overall impression of the model's performance.

The results of the listening audition are next described and obviously this section is the most highly subjective as well as the most potentially controversial. All we can say is that it is a well known fact that reviewers and magazine editors have tin ears (those that is who are not completely deaf) — except, of course, after they have given a product a rave review when the proud manufacturer then praises their perception, knowledge, and even-handed fairness!

The comments on sound quality are unfortunately all too short and may seem rather imprecise due to the use of 'vague' adjectives. However, the introductory section on Listening Tests does explain just what it is we listen for and how we attempt to describe the strengths and weaknesses of a particular design.

The final Verdict can be read in conjunction with our Star Ratings. It is intended as a final summing-up and not as a quick two-line 'Good, Bad or Ugly' definition for those of you who are too lazy to read the preceding page! The Merit rating is a measure of overall excellence and is intended to be absolute based upon all the tests. This does mean that the more expensive products are more likely to be highly rated here than budget priced models, so we have a second Value rating which looks at what is offered for the price. Thus it is possible for an extremely expensive well-designed model to get 5 Stars for Merit yet one for Value and a low-cost model to get 5 for Value and none for Merit. So, often the best buy is the product with three stars on both ratings. Sometimes, of course an expensive model can get a high Value rating — for how do you value a product that stands head and shoulders above the rest?

So it is important that you use the Star Ratings in conjuction with the Verdict.

Finally we have awarded a special distinction to a handful of products where we felt that they represented a particularly fine example of their type. These are not 'Best Buys' in the traditional sense but are commendations for where we feel the manufacturer has achieved a particularly good balance of design, appearance, facilities, performance, and price.

A POTTED HISTORY OF THE HI-FI

We probably owe the present day hi-fi amplifier to the development of talking movies in the 1920's. Although this is difficult to imagine after a trip to the cinema these days, the fact is that before the Second World War the movie industry made all the running in the field of sound recording and reproduction. Much original research work was done (some of which resurfaces from time-to-time when it is 'discovered' by a Japanese manufacturer) and the amplifiers and particularly the loudspeakers were surprisingly advanced. Some 18 years ago (in my youth) I acquired a pre-war cinema horn loudspeaker and 15W valve amplifier (Western Electric) and I can testify that the system could comfortably fill a large hall with sound of a quite acceptable quality (at least for 1965). To my eternal regret I sold this system at a small profit mainly because the loudspeaker took up some 90% of my landlady's living-room! After the war a certain amount of stagnation occured, broken only by the arrival of the tape recorder and the development of higher performance valves as a spin-off of the radio equipment used by the military.

By 1950 some valve amplifiers of good performance were being built using valves such as the KT33C tetrode to achieve 15W output with less than 0·5% distortion. Milestones occured with the design of the Williamson amplifier, recognised as one of the first true hi-fi amplifiers, while the Leak Point One was the first British commercial amplifier to achieve the then magical figure of 0·1% distortion.

The transistor made an undistinguished appearance in hi-fi amplifiers in the early Sixties. Initial designs used transformer coupling between stages and to the loudspeaker, and were noted for their unreliability owing to thermal runaway (instability leading to destruction) and their inability to cope with short-circuits on the speaker lines. The classic transformerless transistor circuit was conceived by an engineer named Lin and, although the original used germanium transistors, virtually all modern transistor amplifiers follow the same philosophy. Some early transistor amplifiers sold well on an image of 'the white hot heat of technology' but in reality they were poor replacements for their valve predecessors. For example, the Leak Stereo 30 was noisy, had high

crossover and intermodulation distortion and sounded rather awful. Yet the preceding Leak Varislope pre-amplifier and TL50 power-amplifier (all valve) were classics of their kind.

The availability of reliable low-cost silicon transistors made possible the design of more successful amplifier designs such as the Quad 33/303 and the Cambridge Audio P40, although as late as 1970 there were many valve amplifiers selling well including the Quad II and Radford STA100. With the transistor came the mass-production amplifier at an affordable price; the recipe for eventual Japanese domination of the market. Back in 1970 there were only some 25 Japanese amplifiers on sale in the UK, these being made by Akai, Nikko, Pioneer, Rotel, Sansui, Sony, Teleton and Trio. By 1975 there were over 100 on sale (excluding receivers) and the rest is history.

Since the early Seventies the transistor amplifier has developed only in what might be termed detail improvements, the basic circuit topology remaining much the same. The commercial availability of power MOSFETS was presented as a new departure in amplifier design, but in practice they have proven to be alternatives rather than replacements for the bipolar power-transistor.

In many ways the development of hi-fi amplifiers has gone through a full circle. In the 1950s and early 1960s, hi-fi buffs were a minority group who took their interest seriously and who usually had a degree of technical knowledge of the subject. Home-brewed modifications to commercial equipment and self-built equipment were common in the quest for improved fidelity. But the late 1960s and 1970s saw an explosion in home audio systems stimulated by the excitingly fresh pop music of the decade and fuelled by the availability of budget priced Japanese equipment and the new-found spending-power of youth. In the 1960s every household had a transistor radio and a valve '3W or bust' record-player, but in the 1970s every home needed a 'HiFi'. The Japanese kept the market on the boil using the proven gambit of planned obsolescence; each and every year a new product range offering a new technology to overcome the (previously unmentioned) limitations of last year's products.

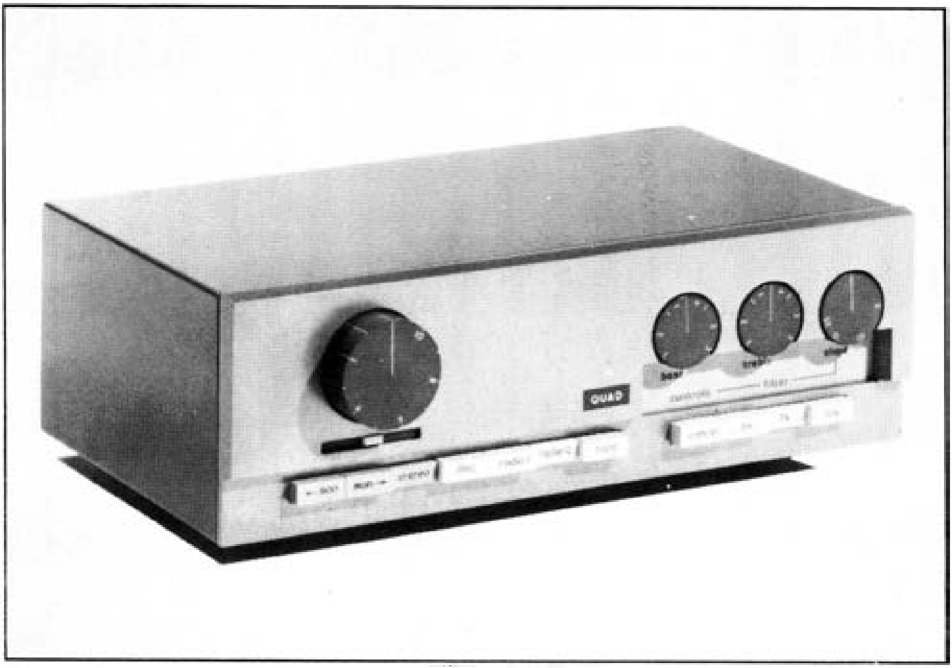
AMPLIFIER

Now the hi-fi market is well past its peak and quite a few manufacturers have gone to the wall. It has also split, with Japanese still needing the diminished mass-market but having to fend-off competition from Taiwan and Korea. The old specification war is back with each company trying to offer more facilities and specifications for the same price. (We've got 30W, power-meters, Dolby, LEDs, Metal tape, etc, etc.)

The enthusiast or audiophile market is still there but has now become more visible with the decline of the mass-market which has now turned to video. This specialised market has given respite and some success to those British manufacturers who have survived, together with a number of fledgling companies. However, the respite may be short for already the signs are that the Japanese are intent upon getting a share of this action as well.

At a higher level there is the 'exotica' with systems such as the Levinson costing upwards of £30,000. In real terms these systems are pretty irrelevant for, although some do achieve high standards of sound quality, they are mostly purchased by doctors, lawyers etc. looking for tax relief on their embarrassingly large incomes.

Indeed one of the more endearing qualities of the modern amplifier is that, provided a competent job is done by the designer, the audible differences between a £300 and a £2000 model can be quite small; certainly not so gross as to bring despair to the enthusiast who has more love of music than he has money.



A famous flashback — Quad's 33 and 303 pre and power amplifier

WHAT DOES AN AMPLIFIER DO?

To most non-technical people their hi-fi system is a mass of wires, circuits and mystery which seems beyond their understanding. But if we take the amplifier as an example the circuitry and switches which you really need (as distinct from want) are very few. Fig. 1 shows the most basic form of amplifier which has but one control on the front panel — the volume control — and three rear panel connections for signal input from the turntable, outputs to the speakers and a cable to the mains supply. Inside the box there are three circuit blocks: the preamp, the power amplifier and the power supply.

In the majority of amplifiers, those termed 'integrated' amplifiers, all these sections are contained in the same box. Amplifiers can, however, be separates and in this case the pre and power amplifers are in separate boxes each usually having its own power supply. Some of the more exotic designs even separate the power supplies into their own boxes.

Because in the case of separate pre and power amplifers there are less physical constraints on the layout and location of the various circuits and interconnections this has tended to be the preferred arrangement for the higher quality products. But the act of integrating both saves the expense of a second case and chassis, separate power supplies and many of the interconnections, so this is obviously the preferred arrangement where cost is the main consideration.

PREAMPLIFIER

The preamplifier block has two main functions to perform. First it must amplify the signal voltage from the few millivolts produced by a pickup cartridge up to the hundreds of millivolts needed to drive the power amplifier. The second function (but only when using disc sources) is to provide the frequency response correction commonly known as equalisation. This is because records are cut with frequency response to RIAA standards (eg. bass cut and treble boost), so upon replay an equal but opposite correction must be applied.

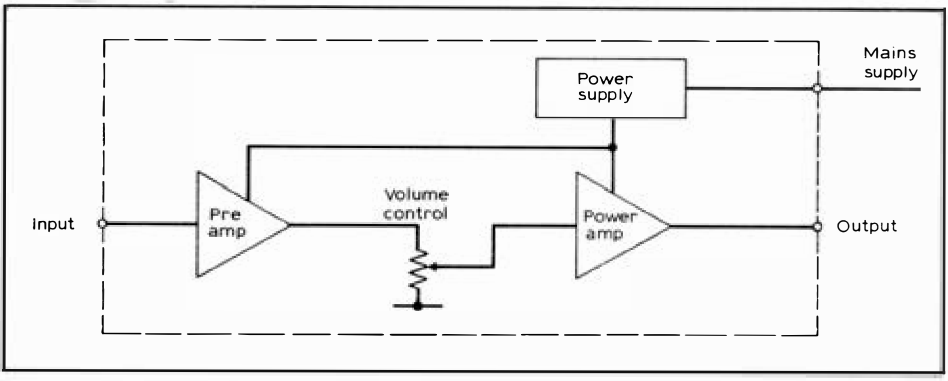
Having fulfilled these two primary aims the preamplifier can be embellished by a number of additional controls and facilities, all of which, except for the volume control, can be considered to be optional.

VOLUME CONTROL

At first glance the volume control is a simple device which does no more than increase or decrease the signal level. In fact it must have some important characteristics. First its law or rate of attenuation must be such that you get a smooth increase in level throughout its travel and not a sudden jump in the first quarter of rotation and little change thereafter.

Secondly the two volume controls (there are two although they operate on a common spindle) should track each other accurately so as to preserve a constant channel balance. Most controls are reasonably

FIG. 1. The most basic audio amplifier.



accurate at the maximum settings but get progressively worse at lower levels. Unfortunately few of us listen at full volume!

Then there is the matter of range. A typical volume control has some 60dB of range (eg 1000:1) and it is desirable that our normal range of listening levels are set in the top half of the control's rotation where the channel tracking is at its best. However the amplitude of the signal before the control really determines its setting for, if at half travel the power amplifier is driven into clipping, the top half of the control is redundant for the amplifier won't get any louder.

Mind you, I do recall that when I played in rock-bands I did know a guitarist who played continuously at maximum volume. Whenever the rest of us complained he would point to the volume control and protest with some conviction that it was "only on number three"!

It is useful to have a −20dB muting switch so that the range of the volume control can be shifted, but better still are the switched gain settings of the Quad 44. The gain of the preamplifier can then be switched to suit different signal sources so avoiding having to turn the volume control to either extreme.

It has been fashionable for volume (and other) controls to rotate in steps to emulate professional attenuators. However, these expensive components are actually multi-position switches connected to ladder networks of close-tolerance resistors to give precise dB steps. The low-cost domestic equivalent is an ordinary potentiometer having a cog-wheel and ratchet. It may feel nice but does have the disadvantage that at low settings the volume does tend to jump in large steps.

BALANCE CONTROL

In a perfect world there would be no need for a balance control and indeed many amplifiers do now omit it. Traditionally it has been useful for balancing up the differences in sensitivities between two loudspeakers but if that is necessary the speakers should be replaced! More reasonably if your seat is not equidistant between the speakers the balance control can be used to decrease the level of the louder channel to maintain a central image.

Furthermore quite a few cartridges have a channel difference in level of 1 or 2dB and this small, but audibly irritating difference, needs to be corrected.

INPUT SWITCHING

So far we have concentrated on the use of the preamplifier for the replay of records but other music sources include the radio and pre-recorded tapes and cassettes. These additional inputs are normally switched in just before the volume control and the sensitivity is such as to match the output from the disc-amplifier. Thus if the preamplifier has a disc sensitivity of 2mV and a disc stage gain of 50 the signal level at the volume control will nominally be 100mV. Then if the radio and auxiliary inputs are made to have sensitivity of 100mV there will be no need for significant movements of the volume control when one input source is changed for another.

MODE SWITCHING

Again this is a switch which has disappeared from many amplifiers. In its simplest form it switches between stereo and mono — when the two channels are commoned. Some amplifiers have more complex switching to give Mono R (both outputs carry the RHC input signal), or Mono L, Stereo, and Stereo Reverse. However most users are unlikely to find much use for such switching.

MICROPHONE INPUTS

A rare one this although it figures in two of our review amplifers, the Teac and the Dual. Usually it gives a mixed mono feed to both channels and so can be used for those 'Sing along with Joe' sessions if you're so inclined. More often it's of interest to those with relatives who want to borrow a PA system for the church fete/social evenings/daughter's birthday disco/etc. So if you buy such an amplifier don't tell anyone!

SPEAKER SWITCHING

This is a very convenient feature but one which is also in some disrepute. All switches, even those (rarely used) ones with gold-plated contacts, do suffer from airborne pollution and oxidiation which results in contamination of the contacts. The connection then becomes non-linear and, in the case of loudspeaker switches, this can cause a ten or even, in severe cases, a hundredfold increase in distortion. As a result most of the better amplifiers use a direct connection to a single pair of speaker terminals. A&R with the A60, and Rotel with their 800 series adopt an interesting compromise. Speaker A is directly wired while Speaker B can be switched.

The effect of two parallel speakers should also be considered from the viewpoint of amplifier loading. A typical 8ohm speaker will represent an impedance of 5ohms over some frequencies. Wire two in parallel and you have 2½ohms — a problem if the amplifier has a limited output capability. So if you are expecting to use two speakers on occasion you should check our 4 and 2ohm output figures with care.

FLASHING LIGHTS

For the main part the use of signal level displays and power meters on amplifiers cannot really be justified. The average listener will know when he's over-driving his amplifier for the distortion resulting from clipping will be quite audible. Nor can they ever tell you more than the level of the output voltage for the power scaling is referred to a constant 8ohm resistance — something a loudspeaker certainly is not.

Three forms of level display are in common use. The good old VU meter on the cheaper models, an LED bar indicator on the majority and the fluorescent (or liquid plasma according to some copywriters) displays used on some higher priced models.

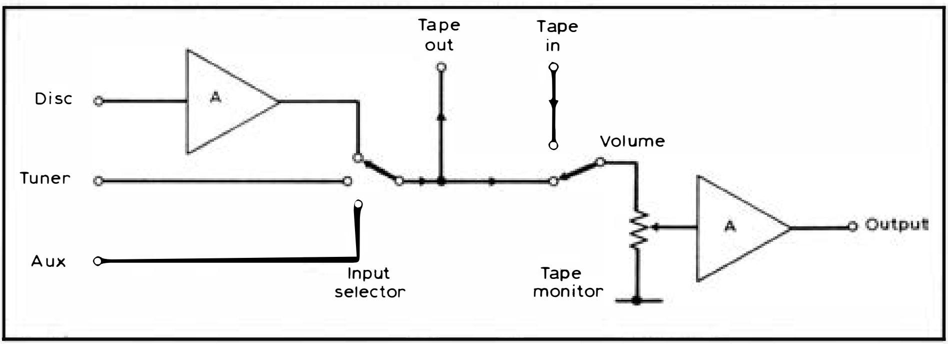
Rather more useful are those indicators which confirm control settings. Thus LEDs or illuminated legends will confirm the choice of input source and whether a tape source has been switched in. Most useful of all on some models is the indicator which shows that the protection system has been activated. Without it the stubborn refusal of the sound to emerge from the speakers can result in much loss of time and temper while leads and connections are fruitlessly checked.

TAPE INPUTS & SWITCHING

For the purpose of making a tape copy of a record (not that you would, of course) or a radio programme, just one tape connection on the amplifier is adequate. Most, though, have two such connections so that copying or, more correctly, dubbing can take place between two tape decks. Fig. 2 shows the most basic tape connection where the tape output comes after the input selection and so carries the same signal as the amplifier. To replay the tape or for off-tape monitoring (using a three-head machine) the tape-monitor switch is used to select the tape source. (Sometimes this tape monitor switch is located after the main volume control so the playback level must be set on the tape deck.)

The provision of two tape connections brings with it the ability to dub from one deck to the other. On low-cost amplifers this can

FIG. 2. Basic tape connections and switching.



only be done from Tape 1 to Tape 2 but on most amplifiers this dubbing is on both directions, eg. 1 to 2 and 2 to 1. With some designs this dubbing can take place without interrupting the main signal, so that you can copy a tape while listening to a radio broadcast.

The best amplifiers (from the tape enthusiast's viewpoint) are those with a separate tape source switch. As shown in Fig. 3 this enables any of the input sources to be fed to the tape deck and another source to be fed to the loudspeakers.

One thing to be wary of is the confusion that exists between DIN levels and DIN sockets. For the main part DIN signals are of low-level and from high-impedance sources and so can (unless you have a DIN matched system) lead to compatability problems. In practice, though, the use of DIN sockets on British and Japanese amplifiers is a matter of convenience or choice rather than an acceptance of DIN standards, so for the most part the signal levels will be the same as those at the equivalent phono-socket.

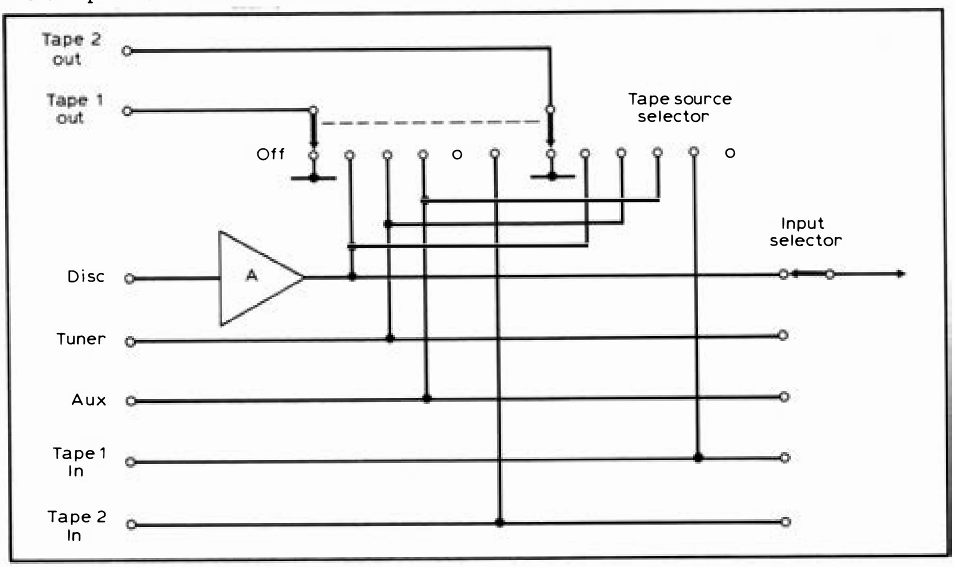
TONE CONTROLS

It is a strongly-held belief that in the ideal world there is no need for tone-controls and so they should be discarded. Now if the cartridge/arm/turntable/amplifer/loudspeaker/room acoustics are all well matched and linear in overall response then this is true. To many people this brings visions of fine tuning followed by careful positioning of the furnishings and an admission not to move anything.

Unfortunately tone controls have fallen into disrepute because of the almost universal use of the feedback type arrangement which can and often does bring as many sonic problems as it cures.

Still it must be recognised that the great majority of hi-fi owners, who have admittedly imperfect systems, will not countenance doing away with tone controls. If an injudicious choice of loudspeaker results in a booming bass in your listening room then a degree of bass cut can be used to minimise the effect. Similarly the use of some cartridge/speaker combinations can result in a 'shreiking' treble which is unbearable. Again the application of some treble-cut is the lesser of two evils. Although both problems could have been avoided by buying the right equipment this is often not possible because of economic restrictions or because of plain bad advice (which is available in unlimited amounts).

FIG. 3. Tape source selection.



The best tone control arrangements are those which feature a tone-cancel or bypass switch so that you can easily ascertain the differences you introduce or, if you are compensating for a cartridge/turntable problem, enable you to quickly restore a flat response when listening to the radio, without disturbing the control settings.

A problem with many tone contols is that an adjustment made, say, to boost the bass, also effects the midband. This is because the response curves are like those in Fig. 4. For many people the curves of Fig. 5 are more use in practice. Magazine reviews can often help but your ears should tell you what you want to know.

One interesting control offered by Lux and Quad is 'tilt' which keeps the response 'flat' but when set for, say, 1dB treble boost has an equal 1dB cut in the bass with the response following a straight line between these points. The effect is to tilt the whole response about the midband without introducing any audible humps in the response (Fig. 6).

Graphic equalisers are now entering a well-deserved twilight period in their development. They have never proved that useful or popular and indeed do not feature on any of the models reviewed in this Buyer's Guide. If fitted they are best switched out of operation except for one possible application. That's the recording of tapes for the car stereo. The acoustics of the average car are so weird that the application of quite great response corrections can often give much improved playback. Basically you are then applying some frequency pre-emphasis during recording and then the car's acoustics provide the de-emphasis. Crude but effective.

FIG. 4. Tone control response curves showing the effect on midrange.

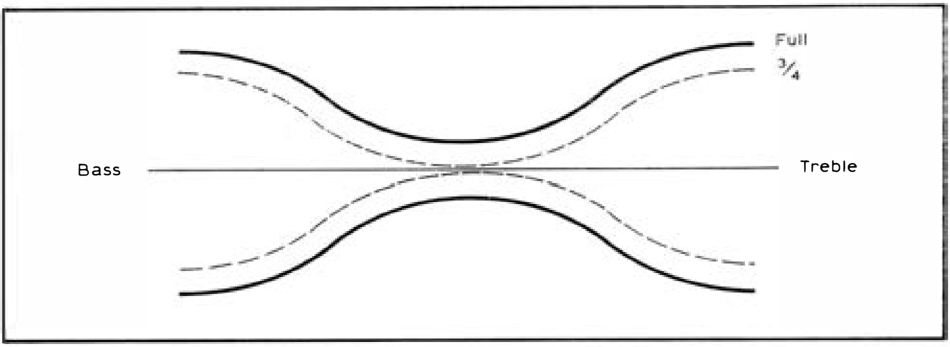
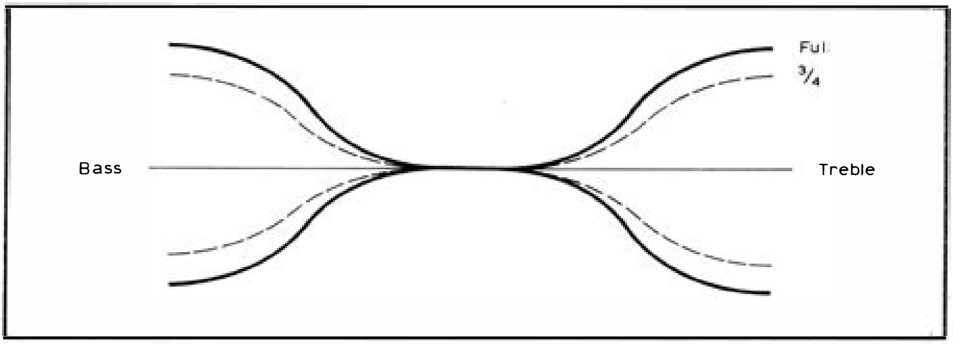


FIG. 5. Tone controls which have less effect on midband frequencies.



Esotec is the crème de la crème.

Shown here is the Esotec stereo pre-main amplifier which offers class A or class AB operation.

In class AB mode it delivers 120 watts per channel RMS of high speed amplification. In class A mode it delivers 30 watts per channel RMS of silky smooth supreme fidelity.

It is of course designed to drive difficult speaker loads.

The control amp section includes a high quality MC head amp designed to complement the best cartridges available today.

Naturally such equipment is not available in every hi-fi store. We make precious few of them but we make each one very, very well.

For further information on Esotec equipment contact us direct, or phone the number below.

ESOTEC SERIES
marantz®

Marantz Audio (UK) Ltd., 15-16 Saxon Way Industrial Estate, Moor Lane, Harmondsworth, Middlesex. Telephone: 01-897-6633/9.

FILTERS

By contrast the use of correctly designed filters can be beneficial. Ideally in an audio reproducing system the useful bandwidth of the programme source should first be ascertained. If we, for the purposes of discussion, say that a record disc contains useful programme material in the band 20Hz to 20kHz then it would seem sensible to restrict the bandwidth of the amplifier accordingly. But the roll-off of an amplifier's response is relatively gradual and close to the roll-off frequencies there are response aberrations such as amplitude ripples and phase shifts. It therefore is prudent to extend the response an octave in each direction to give (in our example) −3dB points at 10Hz and 40kHz

So in effect the amplifier (provided it is not one of the exotic Japanese DC — 10MHz types) starts off with two filters in circuit. Further filters can be switched in, these normally being at both ends of the response (high filters and low or subsonic filters). The high frequency filter should not, in a well engineered system, serve any useful purpose other than to curtail the high level of HF 'hash' generated by moving-coil cartridges. Unfortunately to do this the filter needs to be at the input of the disc amplifier not its output.

The subsonic filter can be useful in avoiding the LF muddle which can result from the resonance of the arm/cartridge combination.

In terms of effectiveness the faster filters usually perform best these being 12dB/octave (or two-pole) or 18dB/octave (three-pole) but they are also the filters which tend to be the most audible in terms of colouration. The 6dB/octave (single-pole) filter isn't very effective but doesn't produce colouration.

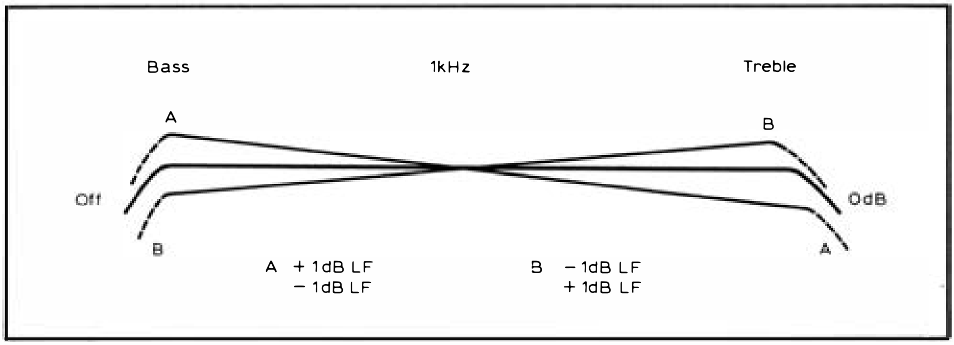
On Japanese amplifiers you will find the ubiquitous loudness filter. This circuit attaches a frequency selective network to the volume control to produce a particular family of frequency response contours. The lower the volume the more the low and high frequencies are boosted relative to the midband. However, such devices invariably give rise to an unnatural sound and are best ignored.

CARTRIDGE MATCHING

Until relatively recently little attention was paid to the problem of cartridge matching at the input of the amplifier, although for years Cambridge Audio tried unsuccessfully to persuade the industry and public alike of its importance. Traditionally the input loading comprised the 'standard' 47kohm resistor and the capacitance inherent in the arm cables (typically 150/200pF). The relatively high inductance of the moving-magnet cartridge formed with this CR load an effective low-pass filter which rolled-off the HF response. So, of course, changing the C and R values will effect the filter characteristics both turnover frequency and rate of roll-off.

One consequence of the work done for CD4 quadrophony was the development of moving-magnet cartridges of extended

FIG. 6. The ideal tilt control.



bandwidth and the adoption of low-capacitance lead-out cables on the arm. Thus to 'tame' the HF response of the cartridge (particularly the tip-mass resonance) may require a higher shunt capacitance and lower resistance at the input of the amplifier.

Some cartridges (Shure and Ortofon come to mind) do benefit from a highish capacitive loading of the order of 500pF but to have any universal application the amplifier's input capacitance should be in the region 100 to 150pF. It can then be increased by using adaptors such as the QED.

Some typical measured values are:

Akai AM-U33	49kΩ + 135pF
A&R A60	49kΩ + 115pF
Denon PMA 510	50kΩ + 300pF
Denon PMA 540	50kΩ + 210pF
JVC AX3	47kΩ + 220pF
NAD 3020	48kΩ + 128pF
Pioneer A7	50kΩ + 218pF
Rotel RC/RB1010	47kΩ + 370pF
Sansui AU-D9	46kΩ + 220pF
Technics SU-V3	47kΩ + 150pF
Trio KA800	33/50/100kΩ + 202pF

POWER AMPLIFIER

Essentially the power amplifier is there to provide current gain although for engineering convenience it is usually arranged that they have some voltage gain as well. In theory the power amplifier should act as a voltage amplifier so that the output voltage remains constant regardless of output loading. This would suggest that the amplifier should have an unlimited output current capability and an infinitely low output impedance but for obvious economic reasons these requirements are compromised. One designer's problem is how to compromise.

Consider first the output current and let us compare two amplifiers:

A with a peak output current capability of 5 amps and B with a capability of 20 amps. If they both have a 100W (8ohm) output their performance will be:

LOAD OHMS	A	B	IDEAL AMP
8	100W	100W	100W
6	75W	133W	133W
4	50W	200W	200W
2	25W	400W	400W
1	12·5W	200W	800W

With the amplifier A the design compromise makes it unsuitable for loudspeaker loads which fall below 8ohms impedance. By contrast B is fine down to 2ohms.

Now consider output impedance which usually rises at high frequencies because of the presence of the output series choke necessary to preserve stability.

If amplifier A has an output impedance of 0·6ohms and B of 0·1ohms consider the voltage across a speaker whose impedance varies from 32ohms to 4ohms.

OUTPUT VOLTAGE

	Max. 32ohms	Min. 4ohms
A	−0·1dB	−1·2dB
B	−0dB	−0·2dB

In the case of amplifier A the sound level will drop whenever its impedance drops.

Many of the problems associated with power amplifiers do stem from the fact that it is all to often modelled on paper as an operational amplifier with feedback and one which provides linear amplification of an input voltage signal.

In practice the power amplifier has to both source and sink the large current demands of the speaker which is a load having variable resistance, capacitance and inductance as well as some nasty characteristics such as the ability to store energy for short periods.

CHOOSING AN AMPLIFIER

First of all, no matter what anyone tells you, there is no such thing as the ideal amplifier. If this were so one company would be making a lot of money and the others would be rushing to make their copies. It's easy for magazine reviews to suggest that you should make sacrifices to buy a Naim or Threshold and none other, but when you've got a wife, kids and mortgage or if you've got an expensive girl-friend and a new Cortina on the HP, it's very difficult to make the necessary adjustments to your lifestyle. For this reason we've largely ignored the 'exotica' in the compilation of this guide.

What most of us are looking for is an amplifier which sounds almost as good as a Threshold; is nicely styled and decently finished; incorporates a good range of facilities; has plenty of power output and retails for £90 including VAT. And, of course, it should never go wrong.

Well when we come down from cloud-cuckoo land we realise this isn't on and we must make some compromises. First we should try to analyse what it is that we want or need (this being the higher state of desire). Unfortunately few of us are good at this sort of logical decision-making. It's all too common for us to develop an overpowering desire for a particular model and then spend the next year thinking up justifications for our purchase even when it is obvious that a big mistake has been made.

If sound quality and accuracy is the overriding concern then you must have the best you can afford commensurate with what you spend on the rest of the system. So, for example, it makes nonsense to blow £800 on an amplifier and be left with a cruddy £79 turntable. If you can take a long-term view it pays to buy the best even if you have to progress in agonisingly slow steps. First the right turntable/arm/cartridge, then the amplifier, and finally the speakers. This way at least you avoid the horrors of the trade-in and the discovery that secondhand hi-fi never quite realises as much money as you expected.

In your journey through this particular jungle you do need an understanding dealer with whom you can develop a long term relationship. This dealer is the sort who will be prepared to forego a bit of short term profit and accept the aggravation that you will undoubtedly cause because he knows you will keep on coming back. How do you find the right dealer? Well his stock is a guide so avoid the guys with fridges lined-up along one wall! Recommendation is the best way, but if in doubt wander in and see how you are treated. I believe in playing the 'Not the Nine O'Clock News' punter who enquires after a new gramophone. If you're treated with courtesy and a concern to establish your true needs then you're probably in the right shop.

For more of us 'good enough' is all we need; a reasonable standard of performance at a price we can afford and a system which will do us for a number of years. Again, though, some thought needs to be given about which compromises we can accept. The amplifier manufacturer can, for a given retail price, only spend so much on the component parts. If he fits power-indicators and 5-band graphic equalisers he must omit components elsewhere — usually in the circuitry that counts towards the sound quality. Equally it is fair to assume that for the same price an unpretentious amplifier having only the bare minimum of features will have more money spent where it counts (of course the manufacturer could be making a much bigger profit but this is unlikely).

If you buy an amplifier which our tests reveal is a poor match for low-impedance of 'difficult' loudspeaker loads then the choice of loudspeaker will be restricted. If you buy an amplifier having no moving-coil input then the purchase of such a cartridge will be accompanied by a bill for perhaps £60 to cover the cost of an external head-amplifier.

At some point the question of power output will rear its head. As our tests show the manufacturer's figures aren't always a good basis for comparison. The PS Audio M2 is rated at 40W and gave 42W, the Hitachi 6800 is rated at 60W and gave 90W. As a general rule you should go for higher than lower power output and pay less attention to the rating printed on the loudspeaker. For short periods most loudspeakers can handle very high powers and indeed are more likely to be damaged by a small amplifier driven into clipping and thus putting high levels of harmonics into the small tweeter.

More important perhaps is the sensitivity of the loudspeaker. For example a loudspeaker with a sensitivity of 86dB (1W at 1 metre) would have to be driven by 200W to sound as loud as a loudspeaker of 92dB sensitivity! An old 'Guy R. Fountain' Tannoy horn loaded enclosure driven by a Quad II 15W amplifier will reach quite deafening sound levels.

Of course, amplifiers having the same or similar power-ratings do not always sound equally loud. There is the problem of protection limiting at lower load impedances (see the section on power amplifier performance) and the fact that some amplifiers can produce a very high short term output when compared to their continuous output. This can be seen by comparing out measured figures for continuous average output and for the short-term (tone-burst) output.

Yet this is only half the story for some amplifiers can work right up to clipping with little audible deterioration in the sound quality, while others seem to sound progressively worse as the volume control is rotated.

So we come finally to the importance of a proper demonstration comparing like with like. After assimilating our comments and those of others you mut listen to the amplifier in the complete system to ensure compatibility. If the amplifier seems incapable of driving your favourite loudspeakers without sonic distress, the fact that it has facilities for three tape-decks and a fluorescent power meter cannot disguise the fact that you have made a bad buy.

EQUIPMENT USED FOR THE TESTS

MEASUREMENTS

All the equipment used for the measurements came from the author's laboratory and included:

Hewlett Packard 8903A	Programmable Analyser
Hewlett Packard 334A	Distortion Analyser
Hewlett Packard 239A	Low Distortion Oscillator
Hewlett Packard 3582A	Fast Fourier Spectrum Analyser
Hewlett Packard 3580A	Digital Spectrum Analyser
Hewlett Packard 85	Computer
Sound Technology 1700B	Sound Measurement System
Radiometer BKF10	Automatic Distortion Analyser
J.J. Lloyd	X-Y Plotter
Dolby Type	CCIR Weighting Filter
Tektronix 2215	Oscilloscope

LISTENING TESTS

Oracle Turntable wth Fidelity Research FR64fx arm

Linn Sondek Turntable with Mission Arm

Shure V15/5, Denon DL103, Keotsu, Technics U205C/3, Dynavector DV100R cartridges

Revox A77HS tape-recorder

Chartwell PM450 speakers

Rogers Studio One speakers

Quad ESL63 speakers

OUR THANKS TO:

Wilmex Ltd	loan of Fidelity Research Arm
Technics Ltd	loan of cartridge
Dynavector Ltd	loan of cartridge
Shure Electronics Ltd.	loan of cartridge

ASSESSING AMPLIFIERS: LISTENING

In some quarters the evaluation of audio amplifiers on the basis of a listening test is considered to be illogical, unscientific and untrustworthy. However, viewed coldly there is nothing illogical about playing music through an amplifier and listening to the results. I can be told that a frozen steak and kidney pie contains all the right ingredients with the correct percentages of added vitamins and yet still prefer the home made efforts concocted by my wife. The fact that there is too high a proportion of kidney to steak will be irrelevant if I prefer it that way. So it is with amplifiers. I know some enthusiasts who listen through full-size JBL studio monitors driven by 350W Phase Linear amplifiers and they accept that the system generates audible colourations and has a far from flat frequency response. They accept these defects because the end result is so satisfying in other respects.

Once it is accepted that a perfect audio reproduction system doesn't exist, it becomes easy to discover which particular combination of virtues and vices is preferred and which isn't. Often such a judgement can be arrived at fairly quickly although the short auditioning time gives rise to suspicions about the validity of the judgement. The real problems come when we try to justify our choices for in our world it is not enough to say "that's the one I prefer"; everyone wishes to know 'why?'. Although there is some overlap between the conclusions formed from measurements and from listening (for example, excessive noise or frequency response errors) the problems occur because there is too little correllation.

On the laboratory bench the amplifier is operating under a defined set of conditions with a known non-complex signal and a set of characteristics is plotted to explain the amplifer's operation in simple electrical terms. But once we install the amplifier in a music reproducing system we lose much of our control. Instead of an audio oscillator we connect a pickup cartridge, and in place of the 1% load we use that most notorious of variables, the loudspeaker. Thereafter a number of interrelated problems can occur. Even if we are experienced and perceptive listeners an error occuring in the ancillary equipment can be of sufficient magnitude to mask a smaller but still relevant error produced by the amplifer. Similarly, cancellation of errors (particularly those of frequency response) can occur. Interactions are always a problem with some cartridges being very sensitive to their loadings, with both response flatness and damping being variable, and many loudspeakers presenting a load which falls outside the amplifier designer's intended range of design loads.

Far more insidious are the subtle interfacing problems which lie in wait for the less experienced and less knowledgeable reviewer. Grounding (or earthing) is a science in itself and although a missing or extra ground is normally audible as a hum or buzz an incorrect grounding arrangement can significantly change the sound quality, particularly if the signal source originates from a moving-coil cartridge. Straight 'hum-modulation' can occur, one of whose effects is to give the perceived sound stage an unstable image, such as to immediately draw criticism from the listener. The presence of 50, 100, 150, 200, 250 – – – – – – Hz harmonics and intermodulation products can also cause a 'thickening' of the sound resulting in a loss of low-frequency definition. It would be very instructive for some reviewers to hear music through a totally hum-free DC powered system and then have one item at a time replaced by its mains-powered equivalent. The difference would, I feel, result in many listeners leaving hot-foot for their nearest garage with a bunch of fivers clasped in their hands!

Equally frustrating, but perhaps better appreciated, are the difficulties that arise when the loudspeaker and amplifier are incompatible. On occasion I have had a particular amplifer A and loudspeaker X as part of my hi-fi system and been quite happy with the quality of reproduction. Then along comes amplifier B which, when substituted in the system, results in an audible drop in quality. The manufacturer then tells me that my loudspeakers X are awful and he's tried them on amplifiers B and C. Many magazine reviews are written as a result of following this sequence which may result in a thumbs-down for the amplifier, the loudspeaker, or both. It is quite reasonable for the reviewer to say that a particular amplifier, when used with a listed system, gave improved or degraded results but it is not logical to say that, therefore, the new amplifier is better or worse than the original

V. MEASURING

when defined in absolute terms.

A particular warning is necessary when dealing with 'system engineered' combinations. For example, Linn Sara loudspeakers sound very acceptable when driven by a Naim 250 amplifier, and so they should being system compatible. But change the amplifier to, say, a Rogers A100 and the Sara's sound is very poor — even unpleasant. Replace them by Rogers Studio One speakers and the sound quality improves. Now it is easy to see that, with two separate comparative tests, in one the Saras would have scored poorly, but in the other the A100 would have been unpopular.

Another problem that rears its head in the course of subjective testing is the 'I might not know much about art, but I know what I like' syndrome. We all suffer from variations on this theme, the most common manifestation being when an artist does a cover version of a successful pop song. The new-release needs to be considerably better than the original before it can achieve any success. If we have a resident hi-fi system at home for some time and we are pleased with the result then it becomes very easy for us to see what we have as 'right'. Then, when another amplifier is used, the differences heard, if there are any, may be classified as 'wrong' instead of 'different'. Blind testing doesn't work here because of the strong possibility the listener will recognise his normal system (particularly if playing familiar records) and subconsciously mark it as the reference. Of course the experienced listener will obviously try to be coldly objective but it is difficult — never more so than when you arrive at the conclusion that a well-regarded and much praised model isn't worthy of its reputation.

So what can laboratory measurements add to our assessment of audio amplifiers? Laboratory measurements are essentially a tool to aid the designer and manufacturer who can lay down a set of specifications and ensure that all the units leaving the production line comply to these and therefore that they all have, for better or worse, identical performance. Some of these measurements can be repeated in reviews without arousing any controversy, eg input sensitivities and impedances, for the results represent useful practical information. But many other measurements need to be examined carefully before any conclusions can be drawn. Gross harmonic distortion (say over 1%) is obviously a bad thing but is there anything to be gained by going from 0·05% to 0·005%? The RIAA equalised response may well be in error by ±0·5dB and this will obviously be audible, but that error realised as a number of 'hills and valleys' in the response will be far more objectionable than a flattish response in the midband rolling-off at the frequency extremes.

Power output is an area in which measurements can be almost as unhelpful as helpful. I recently discussed a new amplifier with a manufacturer who asked what the German DIN rating should be. The amplifier had a rating of 25W, 20-20kHz, both channels driving 8ohms to 0·5% THD. I was asked for a 1kHz one channel 4ohm 1% rating — it came out at nearly 95W! It is important therefore, to specify the exact conditions when measuring power output for any comparisons to be meaningful. They should also be relevant to the sort of load the amplifier will be asked to drive and so it is not enough to just measure the power output into an 8ohm load but also into lower resistances and reactive loads.

If measurements have a failing it is that they don't produce enough information in a form that is capable of correllation with subjective assessments. It is quite possible for a designer to build two amplifiers which measure the same when tested, say, to the IHFM recommendations; but which can be repeatedly identified under blind listening tests. This doesn't prove that measurements are no use — only that we are not measuring enough things or the right things. Over the past few years several new tests have been proposed and regular readers of Hi Fi for Pleasure will be aware that I introduce some of these from time to time. However, it can take a considerable time to implement the test on many products before I can feel confident enough to interpret the results with any great accuracy.

Interpretation is the key to using measurements to assess an amplifier's performance. A modern computer controlled audio testing system, such as the Hewlett-Packard outfit I use, is capable of generating an awful lot of data in a very short time. Most of the routine results are useful only for confirming or otherwise the manufacturer's specification — if they can't get their specifications about right we can't have too much confidence in their other claims. Other measurement results are useful in revealing likely compatibility problems with loudspeakers, cartridges and other signal sources. Finally some tests such as high-frequency intermodulation, non-linear (assymetrical) distortion and peak current capability can give some clues to the amplifier's likely subjective performance.

Measurements are also very useful for revealing faults in review equipment — faults which may not be obvious but which make any auditioning a waste of time. Examples I have found include significant frequency response differences between channels, and excessive harmonic distortion which, while not important in itself, was a sympton of faults in the amplifier. Even an experienced listener is unlikely to spot these faults but he will notice something wrong with the sound quality and comment accordingly.

As things stand at present our measurements are useful because they are objective and reveal the more gross weak areas in an amplifier's performance. Carefully controlled listening-tests do, in the final analysis, confirm whether the amplifier is successful in its role and tend to show up subtle differences which our measurements (or interpretations) fail to pinpoint. But the emphasis must be on the word 'controlled', for it is all too easy to deceive oneself (and others) when making subjective assessments.

The fact that there is still no consistent correllation between the bulk of measurements and the subjective assessment continues to be a source of much frustration. Several designers have shown that adherance to a set of design parameters (interpreted in the form of measured results) can result in an amplifier that will be well liked in listening tests. However the correllation is not so good the other way round when the 'ideal' has been compromised for economic or commercial reasons. It's a matter of threshold. We know, for example, that high levels of two-tone intermodulation distortion do spoil the imaging and clarity when reproducing a complex music signal. IM products at −50dB will certainly result in derisory comments during subjective listening tests while (all other things being equal) products at −100dB will not. But what about in between? In theory it should be easy to take a well regarded 'perfect' amplifier and degrade its performance step-by-step whilst noting the subjective comments. In practice this doesn't prove to be too useful because the interactions that occur result in too many variables.

Finally we come to the problem now being encountered by those who try to evaluate high-quality amplifiers. I have experienced situations where, say, two power-amplifiers offer outstanding sound-quality yet in at least one respect, they do sound different. Which one is right? As things stand it is not possible to answer the question objectively and so a personal preference may well be the deciding factor.

This Buyer's Guide presents both a subjective and objective opinion on the amplifiers submitted for review. But before you buy one of these amplifiers you should, as with all hi-fi equipment, listen to them yourself — preferably with your own ancillary equipment. If you find you cannot live with the sound, it matters not that I have found the amplifier to be the finest I've heard; you simply should not buy it.

LISTENING TESTS

Having explained at length just some of the pitfalls inherent in listening tests, it's obvious that I approached the task of evaluating so many models with more than a little trepidation. As is the case with many experienced listeners, I've often found myself standing up after a brief five minutes worth of music and declaring that I don't like the sound! So far so good, but ask me why and I'm faced with a greater problem. I them need to sit and listen for long periods, working through several passages of music, to try to pinpoint those characteristics and audible symptoms which gave rise to my dissatisfaction.

The auditioning was initially done on a 'blind' basis using groups of, where possible, five amplifiers of similar power rating. One reference amplifier was used to maintain a baseline performance throughout the tests. To avoid any arguments over my choice of reference, I took the egotistical route of using my own Class A system which is not sold commercially. It mattered not whether the amplifiers on test gave a better or worse performance than the reference — only that I should have a constant standard to work against over a long period of time.

Each amplifier was then listened to individually with many pieces of music in an attempt to come to terms with its subjective character. As I was only prepared to listen fully to one product per day this resulted in the project stretching out somewhat, causing panic to the publisher who'd already booked time on the printing presses!

One immediate problem was the choice of ancillary equipment used to build up a reproducing system centred around the amplifier. The choice of turntable/arm/cartridge was a problem because I sought not only a fine programme source but also wished to assess whether the amplifier was compatible with popular cartridges. So, as the main programme source I used my Oracle turntable fitted with Fidelity Research FR64fx arm and the choice of Koetsu or Dynavector Karat Ruby moving-coil cartridges. For compatibility tests, as well as listening to music, my Linn Sondek was fitted with a Mission arm and a choice of Shure M97HE and Technics U205C/3 cartridges. Original recordings were played back at 15ips on my trusty Revox A77HS to give an alternative signal source.

Worst of all was the difficulties surrounding the choice of loudspeaker. Experience had taught me that it was dangerous to rely upon the results with just one loudspeaker model, for an amplifier may be incapable or correctly driving one model yet give a highly creditable performance with another and so be worthy of a suitably qualified recommendation. After much thought I came back to the three models I normally use for reviews. The Quad ESL-63s were chosen for their midrange accuracy, imaging, resolution and the fact that they represent a moderately difficult load at high frequencies. The Rogers Studio Ones were chosen as a typical low-colouration ported two-way system of medium efficiency. And, for their immense power handling capability and extended low-frequency response, I chose a pair of the big Chartwell PM 450P Studio Monitors.

Most of the listening took place in my fairly solidly built Elizabethan home in a room measuring about seven by six by three metres (high), and which has a slightly live character — another factor in the choice of loudspeaker.

For the most part, the subjective assessment of each amplifier represents the views of one man and his dog, but the comments of colleagues and my wife were taken into consideration. This last comment is of great interest for, in common with some other reviewers, I have found that some women are able, despite an apparent disinterest in the task of appraisal, to put their fingers exactly on the problem area. Throwaway remarks like "doesn't that tambourine sound tinny" are often far too accurate for comfort!

Finally, to fill in the background, what about the choice of music? Well I certainly don't intend to print a boring list of records because I can't remember them all. Mainly I play those records I enjoy listening to rather than those records that are designed to impress. As a result most of my records are normal 'off the shelf' pressings rather than direct cuts etc. However, some of my old favourites have been replaced by the 'half-speed mastered' pressings which are now available.

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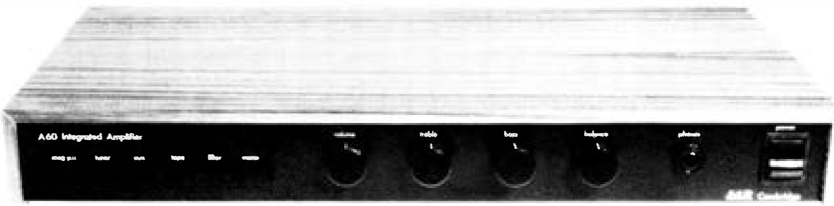

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SUBJECTIVE ASSESSMENT

The great majority of magazine reviews are now subjective assessments but, whilst they are often valid attempts to describe the performance of the equipment, they sometimes leave the reader confused because of the often imprecise description of the sound quality. Perhaps the most famous review comment was the description of "a subtle lack of ambience". Unfortunately it is often difficult to be precise in a short summary of an amplifier's subjectively assessed performance. So, to help the reader I have tried to describe just what characteristics we are listening for.

First of all, I listen to an amplifier's overall tonal balance — whether it seems to have a flat response or has a degree of emphasis of part of the frequency band. Thus, an amplifier with a treble emphasis will sound bright.

Next comes the matter of integration. Do all the components of the sound form a cohesive image or do the different parts of the frequency spectrum seem to separate apart? A simple example would be a vocalist singing a scale of two octaves. Does the sound location appear to move closer or further away as the notes rise in frequency? If they do, then the sound isn't fully integrated and the effect on more complex music can be imagined.

The matter of colouration or distortion is then assessed. This is, for me at least, most clearly apparent on recordings of solo female voice and percussion instruments such as cymbals and tambourines. I listen for the mix of harmonics whose makeup is critical in reproducing these sounds accurately. So, for example, if a cow-bell should go 'tinning' it should not be reproduced as 'tinnnnzzz'.

The next thing to consider is the amplifier's control over the loudspeaker. To many people a lack of control is most apparent when bass sounds are reproduced. Thus a big bass drum will produce a 'bap' sound rich in higher harmonics but, if it is reproduced (assuming the loudspeaker is an accurate transducer) as a 'boom', then obviously the amplifier hasn't exerted sufficient damping of the loudspeaker cone and hence it isn't under control. A similar problem can be heard at mid and high frequencies although it is harder to pinpoint.

Some amplifiers seem to reproduce and, indeed, separate out detail far better than others. This may be a case where you discover for the first time that the two voices singing in harmony are actually three! Often low-level detail is revealed. Some amplifiers seem to reproduce sounds of lower and lower level until at some threshold all quieter sounds disappear. The more 'informative' amplifier continues to reveal sounds even when they are down in the residual background noise.

Such amplifiers are often able to reproduce sounds with greater clarity. Again a good test is female voice; I listen to the diction. With some recordings it is difficult to hear all the words clearly, but the 'informative' amplifier will often reproduce the vocal sounds more cleanly and therefore make the music more intelligible.

Next I listen and try to synthesise the sound stage as reproduced. Where does it extend from and to where? The locations of the instruments and performers should be as precise as the original recording technique allows (the imaging), and there should be a sense of width, depth and height. Most important of all, the images should be stable and shouldn't show a tendency to move whenever the sound level changes. Obviously the choice of loudspeaker is critical for making these imaging judgements but I found the Quad ESL-63s to be admirable in this respect.

Finally I like to consider the dynamic performance. Are the dynamics of the music reproduced accurately and effortlessly, or does the amplifier appear to apply unnatural compression or, more rarely, expansion to the sound? Many amplifiers seem to change their overall sound quality at differing playback levels even though they are not being driven to significant clipping. This characteristic should also be assessed.

Within the foregoing list can be found a complete description of the amplifier's subjective performance and, although some reviewer's comments may be worded differently, their findings can be traced to one of the characteristics I have described.

I did originally intend to publish marks for each of these subjective headings but space, time and controversy defeated that plan. However I do hope that you, the reader, are now better equipped to understand the relevance of my comments.

At all of our branches we hold a superb display of top quality equipment at competitive prices. Demonstrations are available (bring your own record or tape if you like) and our helpful staff are always happy and able to offer knowledgeable, unbiased advice. We can also arrange home demonstrations and installations — just ask for details.

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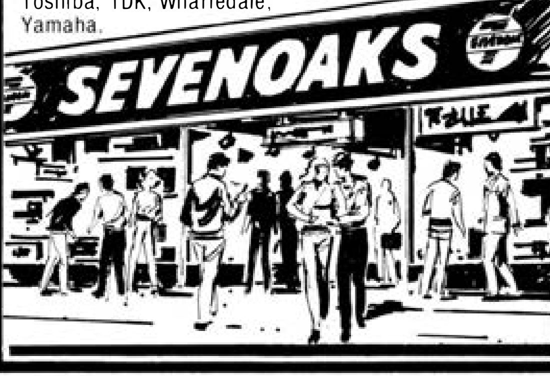

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MEASUREMENTS

REFERENCE LEVELS

To ensure a consistent baseline for all the measurements both input and output signals are referenced to the IHFM recommended levels.

The reference output levels is 2·83 volts which is equivalent to a power of 1W in an 8ohm load (or 2W into a 4ohm load).

Several reference levels are used for the input signal, the largest being 500mV for line (aux, tuner, tape) inputs. The moving-magnet input reference is 5mV; that for moving-coil inputs being 0·5mV (eg. 500 microvolts).

In all the measurements made the signal levels are normally quoted in dB referred to one of these reference levels. The exceptions are power outputs and input sensitivities which have been quoted in watts and millivolts respectively for clarity as these are the units with which readers will be the most familiar.

POWER OUTPUT

This is a measurement of the maximum continuous average power output that can be delivered to the specified load by a sine-wave test signal of frequency between 20Hz and 20kHz with less than 0·5% total harmonic distortion.

The amplifiers were tested with a single channel driving first an 8ohm load and then a 4ohm load. The power output was then measured with both channels each driving an 8ohm load. (Many of the amplifiers were also tested with one channel driving a 2ohm load but the result has not always been quoted.)

The amplifiers were then tested with a 20kHz sine-wave test signal driving a capacitive load (4ohms in parallel with a 2μF capacitor), the output voltage being increased until the total harmonic distortion reached 1% or the protection circuits came into operation (or in some cases until the fuse blew). This maximum undistorted voltage was recorded and can be compared with the maximum undistorted voltage generated across a 4 ohm resistor alone.

A further measurement was made to measure the maximum transient output capability of each mode. A 20 cycle tone burst of a 1kHz since wave was driven twice per second into an 8ohm load and increased in level until visible clipping occurred.

HARMONIC DISTORTION

This is a measure of the total harmonic distortion (including noise but excluding hum) of a sine wave test signal measured at the output of the amplifier when both channels are each driving an 8ohm load at the manufacturer's rated continuous average power output. Measurements are taken at 20Hz, 1kHz and 20kHz and the results quoted in a percentage form.

The signal source used was the oscillator section of a Sound Technology 1700A which has been optimised to generate under 0·001% distortion of the test signal. The distortion measurements were made simultaneously on the 1700A and the latest Hewlett Packard Type 8903A Programmable Analyser. In each case the lowest recorded result was recorded.

Whenever it was suspected that the amplifier under test was generating no more distortion products than existed in the test signal then the analyser output was fed to a Hewlett Packard 3580A Digital Spectrum Analyser and the harmonic structure compared to that of the test signal. When examined in this way it must be remembered that a reduction of the amplitude of a harmonic represents distortion of the signal as much as does an increase. Often such distortion 'cancellation' can lead us to record a falsely low distortion figure.

DAMPING FACTOR

The damping factor is measured at three frequencies: 20Hz, 1kHz and 20kHz and is the ratio of an 8ohm load to the amplifier's output impedance.

eg, DF = $^8/_{Z_0}$

The output impedance is measured with the amplifier delivering an ouput of 10 volts RMS with no load applied. A 4ohm load is then applied and the reduction in output voltage measured on a very accurate Fluke digital voltmeter.

Using this test method some amplifiers can give a false result for their output impedance so an alternative technique was sometimes used.

The output of the amplifier under test was connected via a 16ohm precision resistor to the output of a large power amplifer. A test signal (at 20Hz, 1kHz and 20kHz) was fed from this power amplifier which now represented a current source. By measuring the differential voltage at either end of the precision resistor the amplifier's output impedance could be computed.

NOISE

The signal-to-noise ratios of the disc and auxiliary inputs were measured using the IHF recommended procedure. The applicable reference level signal (see introduction) was fed to the input and the volume control rotated until the output level rose to 2·83 volts across an 8ohm load (equivalent to 1W) this level being taken as 0dB. The test-signal was then removed and a dummy input load substituted; this being 1kohm for the auxiliary inputs and a shielded cartridge for the moving-magnet disc inputs.

The residual noise was measured with a CCIR filter (Dolby Type) and an average reading meter, the result being expressed in dB.

A check of the residual hum level was made using a spectrum analyser so that the harmonics of the 50Hz hum could be isolated and measured.

DISC INPUT OVERLOAD

The overload margin was checked on the disc inputs by steadily increasing the test signal while reducing the volume control setting until the distortion of the input stage reached 1%. The test signal level was then referred to 5mV (or 0·5mV for a moving-coil input) to give an overload level in dB. This figure was checked at 20Hz, 1kHz and 20kHz (with level adjustments to accommodate the RIAA equalisation) so that the linearity of the disc amplifier could be evaluated.

RIAA EQ. ACCURACY

The frequency response of the amplifier was measured via the disc input in order to ascertain the accuracy of the RIAA equalisation. This is commonly checked by feeding the test signal through an inverse RIAA network and performing a swept response onto a pen-recorder, but this was not found to be sufficiently accurate nor could an inverse network be made that was more accurate than 0·03dB from 20Hz to 20kHz.

So instead a computer was used to calculate the frequency response very accurately from the RIAA time constants of 3180, 318 and 75 microseconds. Then, using the Hewlett-Packard Type 8903A Programmable Analyser, numerous fixed frequency measurements were made and the computer (a Hewlett-Packard Model 85) printed out the deviations from the theoretical value. The errors were then plotted point by point onto a graph to give a composite response from 20Hz to 20kHz between limits of ± 1dB.

DISC INPUT CROSSTALK

A 1kHz test signal of 5mV level was fed into the disc input and the volume increased to give an output of 2·83 volts (the 0dB reference level). Then the input to one channel was removed (the input being terminated by the dummy cartridge load) and the residual 1kHz signal measured to give a figure in dB down from the reference level. The measurement was repeated at 20kHz using a test-signal level of 50mV.

PEAK CURRENT

This test was developed to establish the peak output current capability of the amplifier without it being blown up. (The thought of 50 angry manufacturers was too much to contemplate.)

The output current could be measured by connecting the amplifier to a 0·1ohm resistor and then increasing the signal level until either the fuses blew, the protection circuits operated, or the output stage burnt out. But this would not reveal the transient capabilities of the amplifier so important in, say, reproducing a kick drum.

So the amplifier was used to charge and discharge a large high energy non-polarised capacitor when fed with rectangular pulses at a 1kHz repetition rate. The photographs show the resulting integrated waveform and it will be seen to have a low energy content and hence a low-heating effect. Examples are also shown of some of the amplifiers tested.

The figure recorded is the peak current and can be converted to an RMS figure by using a 0·707 factor. Then the maximum power-output into a load can be calculated thus:

Let I peak = 5A
then I RMS = 3·5A
maximum voltage across 2ohms
= 2 × 3·5 = 7V RMS

$$\text{Maximum power output} = \frac{7 \times 7}{2} = 24{\cdot}5\text{W}$$

(assuming the power supply and available voltage rails make this possible).

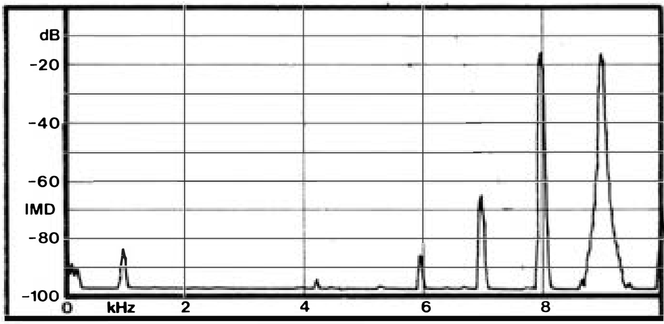
INTERMODULATION DISTORTION

This was a two-tone test using sine wave signals to establish the linearity of the amplifiers at high frequencies. It has been common to measure IM distortion at low signal levels but in addition we also measured at a level 50% down from clipping of the two test signals (this in fact is the level on the published plots although the results of the low-level plots were also considered).

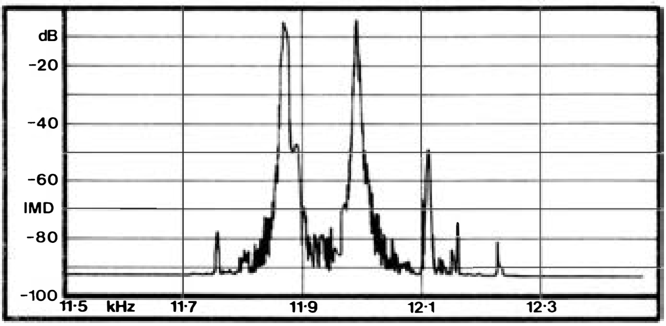
The first test used signals of 8 and 9 kHz and the frequency analysis was performed by a FFT type Digital Spectrum Analyser over a 0 to 10kHz bandwidth. Of particular interest were the IM products which are found at the left hand side of the plots with 1kHz being the main difference frequency, although other IM products would show to either side. Below is a plot of a poor amplifier as an illustration although none of the review amplifiers were this bad.

IM products will also form around the skirts of the test-tones but these cannot always be correctly analysed using an FFT analyser because of sampling and averaging errors. Instead we used a high quality swept analyser (Hewlett Packard Model 3580A) to analyse two tones only 100Hz apart. This is a severe test and all the amplifier plots revealed some sidebands but again an extreme example is printed below for guidance. Unfortunately some of the detail may well have been lost through the act of reducing the plots to fit the pages.

To spare blushes we will not be publishing the identity of the amplifier used as an example!



IM plot of a moderately non-linear amplifier showing the presence of IM distortion products



IM plot of a moderately non-linear amplifier showing the presence of major and minor side products

AMPLIFIER DEFECTS

It is very difficult for us to make a realistic evaluation of an amplifier's potential reliability. Obviously some amplifiers are so badly built that we are surprised they have survived the delivery journey to us in one piece. Fortunately such products are rare, although one of the pre-amplifiers sent to us for this guide was in that class. It has not, however, appeared in these pages because the manufacturer went bust. No doubt there's a moral there! Some amplifiers do suffer from transit damage; we had one out of 50 — 2% isn't a bad average but the reader should be shielded from this problem by the dealer who should at least open the carton and check the unit.

For manufacturers the worst problem is the incidence of random component failure. This can be minimised by the use of good quality components that are working within their ratings. For example, you don't attach a 25 volt capacitor to a 28 volt supply line or run a ¼W resistor at an average dissipation of 1W. A burn-in period at the factory is also helpful because the highest failure rates are early and late in the life of the components. British manufacturers tend to adopt the latter course and rightly so because British and some American components suffer significantly more failures than their Japanese equivalents. The Japanese, though, tend to place their reliance on high component quality, because their products tend to fly-off the production line at a high rate of knots and the thought of a 72 hour burn-in period is the sort of bottleneck nightmare most production engineers dread.

Of the 50 or so amplifiers submitted for review we had minor problems with four. The Hitachi 3800 suffered a broken volume control knob in transit but this was a minor irritation and hardly the fault of the manufacturer. Electrical faults occured in the A & R A60, the Tandberg preamplifier and the Sansui AU-D9. In each case the faults seemed to have been the sort of random transistor failures which could afflict any model. It's just curious that these failures occurred in top-quality models from manufacturers who care about quality control — while all the low-cost products proved to be trouble free.

This 6% failure rate was higher than expected but probably not too far off the truth. Of course, our test procedure might be thought to be more arduous than normal domestic use, but at no point were the amplifiers stressed outside their design limits.

However, amplifiers do fail in use and sometimes this can lead to expensive repair bills. More often the fault is no more than a blown fuses as a result of a supply surge or a momentary shorting of the speaker lines. Such faults are easily repaired by the replacement of the 5p fuse (but *always* with exactly the same type and rating as the original). Unfortunately for the average owner the manufacturers now tend to fit all fuses, including the frequently blown loudspeaker fuses, inside the amplifier which is then boldly printed with the words "No user replaceable parts inside — refer to qualified technicians".

The fuses can be wired into several parts of the circuit and often a bit of simple experimentation can narrow down the fault. If you switch the amplifier on and no indicators light up and there's no faint hum from the power-transformer you know that it's probably the mains supply fuse that has gone. If the supply indicator lights up but there's no output from the headphone socket or from either channel, the chances are that a low-voltage power-supply fuse has blow. Finally if only one channel fails but you have a signal at the tape-out socket and, possibly, at the headphone socket then its a good bet that the speaker output fuse has blown.

When an amplifier does fail it can put DC (eg, one of the supply lines) onto the loudspeaker and you will hear a very loud buzz from the speaker — albeit not for long because it will soon burn-out. Some amplifiers have relays or triac circuits (eg, the Quad 405) to protect the loudspeaker, while others just rely on the fuse. But the fuse can often be poor protection. The average 35W amplifier will have a 4A fuse which requires a sustained minimum current of 8A to guarantee fusing. When the amplifier fails one of its, say, 36 volts supply lines will be across the speaker which if it has a 6ohm DC resistance will be fed a current of 36/6 = 6 amps but the fuse won't blow. Reduce the fuse to 3·15 amps and you're probably safe.

The fact is that the repair cost of consequent failure of the loudspeaker can often dwarf the cost of repairing the amplifier.

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A&R A60

Now getting rather long in the tooth, the A & R A60 has been selling well for the past 5 or 6 years. It started out as a budget-priced model offering fairly basic facilities but the present price is undoubtedly high compared to its imported competitors. The styling is pleasantly simple with a narrow black fascia screen-printed in white and a wrap-round walnut veneered lid. On the front panel the mains switch is dominant and really looks well out of scale with the other controls. The four control knobs are not clearly identified at a glance so some familiarity needs to set-in before you can be sure of grabbing the right control. Nor are there any centre detents or panel graduations to ensure an accurate 'zero' position for balance and tone controls — particularly in the case of the latter as there is no 'tone-cancel' switch.

What you do have is three inputs and one for tape, a mono-stereo switch, and a quite effective high frequency filter turning over at about 7kHz with a 12dB/octave slope

The loudspeakers can be connected 'directly' (although there is a fuse in the output line) using one pair of 3mm sockets or, using the second pair of sockets, via the muting switch on the headphones socket. All input sockets are 5-pin DIN types which many of us find inconvenient, but work I did some years ago (and reported in a HFP article) showed that with its self-cleaning silver contacts the DIN socket gives a far better contact than the oridinary grade of phono jack. A & R also make a moving-coil head-amplifier (see that section) which plugs into the disc input socket and gains its power from the A60. Finally, A & R offer a range of input matching modules which plug-in to give alternative loadings for the disc input.

CONSTRUCTION & TECHNOLOGY

The internal construction of the A60 is tidy with all the components, except the toroidal transformer, being mounted on one large printed-circuit board. This form of assembly, with both the inputs and outputs being virtually direct-soldered onto the board, does give a consistent and predictable standard of performance. The board is screen-printed with all the component designations and carries a number of components accessible to the owner. These are the fuses for the power supply and loudspeaker lines (the latter fuses may blow if some low-impedance speakers are driven hard); and a power supply link to permit the use of the optional HA-10 moving-coil head-amplifier. Finally, near the input sockets there is a plug-in module which carries the input loading components (resistance and capacitance) for the disc-inputs. A range of inexpensive alternatives is available from A & R, or you can make your own.

Through holes in the baseplate you can adjust two small preset controls which vary the tape replay level. They are uncalibrated but can be useful in matching the tape-monitor level.

The circuitry of the A60 is extremely simple incorporating no 'technological breakthroughs'. The power amplifier has the usual differential input stage run at a constant current and the feedback is AC coupled. The output stage is quasi-complementary and uses a pair of 3055 power transistors — slow but rugged.

£198 MERIT ★★★☆☆
VALUE ★★★★☆

One of the most significant aspects of the A60's design is the use of built-in filters to tailor the response. Thus the preamplifier stages incorporate a high-class filter which rolls-off low and subsonic frequencies, while at the input of the power amplifier there is a passive filter which rolls-off the high frequencies and in doing so limits the slew-rate of the incoming signal to prevent any TID. The overall result is that the signal bandwidth is limited to that which the amplifier can process in a linear fashion.

TEST RESULTS

The power-output comfortably exceeded the rated 35W (8ohms) by reaching 48W with both channels driven at 1kHz. The power increased into 4ohms but the A60 was less happy with the capacitive load which couldn't be driven much above half the available output voltage.

Both the harmonic and intermodulation distortion figures were low enough, although not as low as many more recent designs

The preamplifier section measured well with a healthy overload margin on disc and good stereo separation in the midband, although this fell at high frequencies.

The frequency response measurements revealed that the A60's bandwidth is sensibly curtailed to just beyond the audio band and then smoothly rolled-off. The RIAA accuracy was reasonable but showed a slight mid-range emphasis.

In all this amplifier did not display an outstanding performance in any one area but did offer a very balanced specification with no obvious areas of weakness.

The A60 was well-liked during the listening tests, its rendition of the dynamics in musical passages being particularly fine. This amplifier always seemed to be in control of the loudspeaker and in consequence the bass was both detailed and natural sounding.

Tonally the amplifier seemed a little bright yet lacked both the smoothness and transparency of some better designs. The high frequencies were a problem area with some transient signals sounding rather 'splashy' and imprecise on occasion.

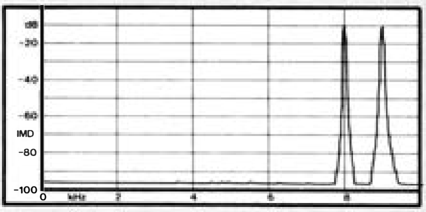
But overall a very satisfying impression was created.

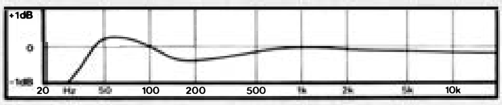
THE VERDICT

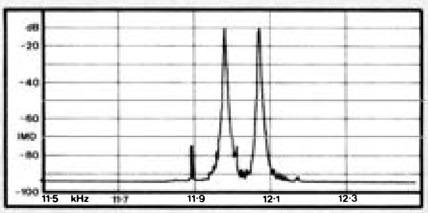
A rather basic amplifier, in a basic case, and having only basic facilities; yet an amplifier that produces a satisfying sound quality despite its seemingly pedestrian specifications. The A60 may be getting a bit long in the tooth and rather more expensive to purchase than its 'budget' image suggests but, when it comes to reproducing music, it still offers good value.

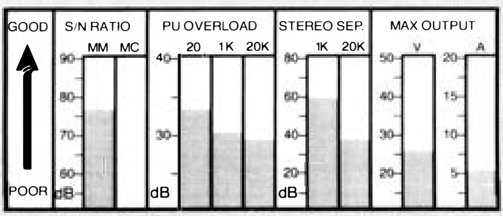
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	50	52	49
(WATTS)	8+8Ω	42	46	40
	4Ω	69	76	68
HARMONIC DISTORTION (% @ 35 WATTS)		0·031	0·027	0·074
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 36Hz		HIGH 24kHz





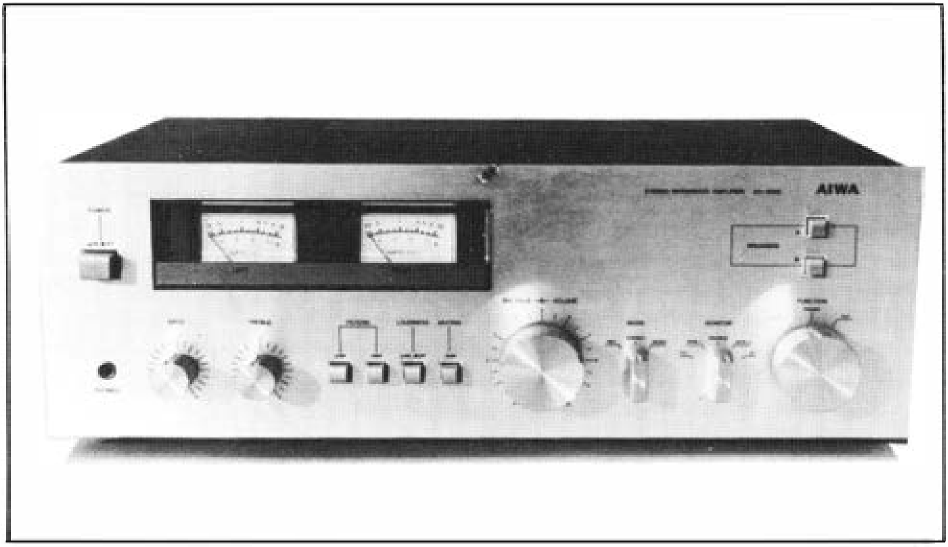




AIWA AA-8500

Aiwa are well established in the hi-fi world and indeed they are a sister company to Sony. In recent years, though, they have tended to concentrate their marketing on rack systems although most of their products are available as separates. The AA-8500 is here reviewed in its 'separate' role as a 50W per channel integrated amplifier. The overall styling is rather odd as it stands, but looks quite smart when the amplifier is rack mounted in combination with a tuner, cassette deck, etc. The front panel is satin anodised with subdued grey printing but split by a blue acrylic indicator strip. The row of selector switches are also in blue plastic and illuminate when pressed — the overall result being quite effective if you like that sort of thing. Above the selector switches is a horizontal slider control for volume, while alongside is a miniature rotary balance control. Indeed the balance and tone controls have knobs so small that it is difficult to use them smoothly.

Four main inputs and one tape connection are provided so tape-dubbing is only possible by using one of the auxiliary inputs. Subsidiary controls include loudness, muting, stereo mode, subsonic filter and a curious switch called DSL. This stands for 'Dynamic Super Loudness' which appears to boost the low-bass in some unexplained way to compensate for the low-bass roll-off of the loudspeaker. Two loudspeakers can be connected and separately switched and the power fed to them is displayed on a horizontal bar-meter (behind the blue strip) which uses seven segments common to both channels.



CONSTRUCTION & TECHNOLOGY

Internally the Aiwa is quite neatly built with some substantial components unexpected in such a 'budget' product. For example the extruded heatsink is very substantial, the transformer is an efficient but expensive toroidal type, and decent heatsinking has been provided for the rectifier and power supply regulators.

Other nice details included the use of high-temperature insulating sleeving on all the mains wiring, and the better quality sealed preset resistors on the boards. The bulk of the circuitry is on one large PCB which is neatly laid out but doesn't carry component designations.

No circuits were provided but it was apparent that most of the amplification is provided by ICs., the power amplifiers again using the increasingly familiar STK series of power-block ICs.

£140 MERIT ★★★☆☆ VALUE ★★☆☆☆

TEST RESULTS

There was litle difference between the single and both channel output power into 8ohms, this suggesting that the power supply is of good quality and 'stiff'. However, the power output into 4ohms only increased by a few watts because of early current limiting which also restricted the 2ohm power to under 34W. The reason was clear when the peak current capability was measured only 5½amps being possible before limiting occurred.

Most of the other measured results were fine although the frequency response was rather unnecessarily extended at the top end, the response being only 1dB down at 150kHz. The stereo separation was fine at 20kHz but could be usefully improved at 1kHz.

Finally a most extraordinarily accurate equalisation response was measured particularly in view of the Aiwa's price. From 30Hz to 20kHz the response was accurate to ±0·04dB.

The sound quality of the Aiwa was quite fair at low levels but when moderately loud the sound became muddled and confused. The bass, never seeming to be particularly well controlled, also became much worse as the sound level increased. Tonally the balance was fairly level but a harshness arose in the treble and so at reasonable listening levels the sound took on a 'boom-crash' quality. More positively the sound was quite lively and exciting so this amplifier could be successfully partnered with a smaller pair of speakers not having much bass extension.

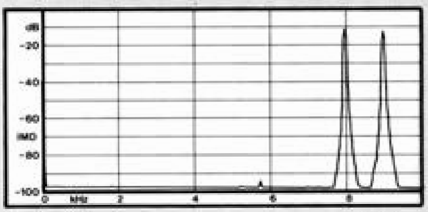
THE VERDICT

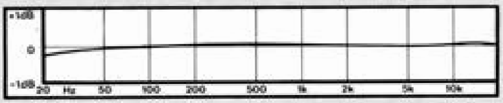
If we look at the all round performance of the 8500 it is difficult to see it as much more than undistinguished 50W model. The price is about average for the power rating and facilities offered. In practice you are unlikely to find the Aiwa particularly better or worse than much of its competition.

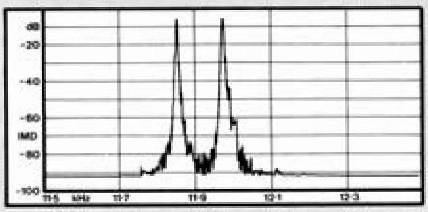
The sound quality is not outstanding but may appeal to those who like a lively exciting sound and are prepared to live with the deficiencies found.

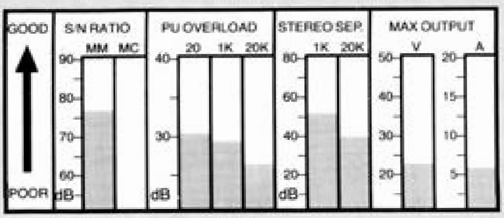
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	64	68	64
(WATTS)	8+8Ω	58	63	60
	4Ω	75	77	77
HARMONIC DISTORTION (% @ 50 WATTS)		0·008	0·003	0·037
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 7Hz		HIGH >150kHz







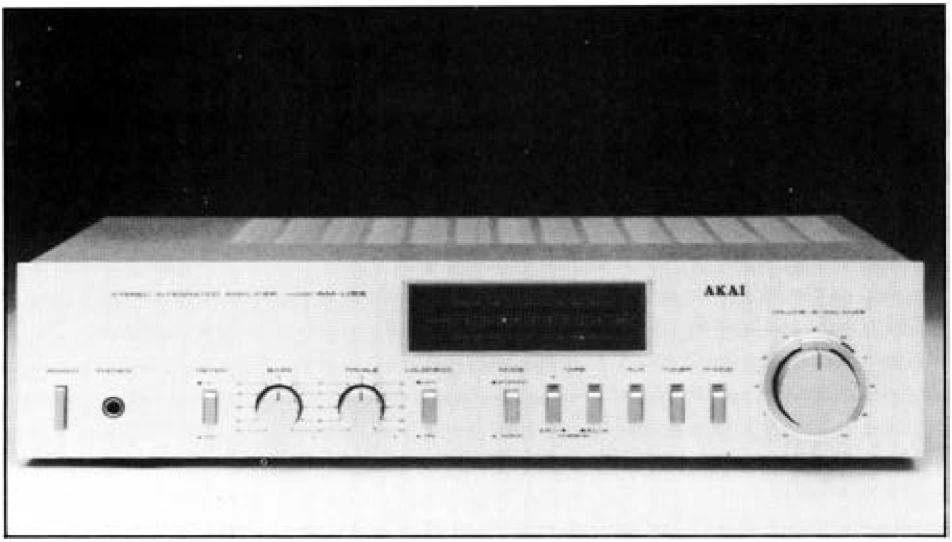


AKAI AM-U33

The Akai AM-U33 is one of a range of amplifiers which differ mainly in terms of their rated power output. The particular model is inexpensively priced but is rated at 43W into 8ohms. The styling is fairly nondescript with an extruded aluminium front panel and a creamy-grey painted case. The panel layout is neat but dominated by a large 'power-meter' bar display using nine LEDs for each channel. The calibration reflects the equivalent power fed into a constant 8ohm load (which a loudspeaker definitely isn't) with two ranges being switch selected — 50mW to 70W and 5mW to 7W.

The volume and balance controls have been made concentric, the large knob being for volume whilst an inner knob with a raised pointer controls the balance. The two controls do not interfere with each other so there is little likelihood of upsetting the channel balance while rotating the volume control.

Inputs are provided for a moving-magnet cartridge, two lines and two tape decks and a small LED indicator is positioned above each selector switch to confirm the setting. The amplifier has bass and treble tone controls (which cannot be bypassed) but no filters. Only one pair of loudspeakers can be connected and these are switched out when a pair of headphones is plugged in. Finally a potentially useful feature has been provided in the form of separate preamp output and power-amp input sockets connected normally by a jumper link. Thus it is possible for the owner to upgrade either the power-amp or preamp section at a later date or to use the AM-U33 as a slave amplifier.



CONSTRUCTION & TECHNOLOGY

The AM-U33 is substantially built into a plated steel chassis which has screening panels around the preamplifier circuits. Most of the circuitry is mounted on three large circuit boards linked by ribbon cables terminated by plugs and sockets for ease of service. Particular emphasis has been placed upon electrical safety, so all the mains wiring and connections are sleeved and shrouded. A 3-core mains cable is provided which is connected through an IEC pattern plug and socket. To judge from the overall internal appearance and the components used it is clear that Akai have not skimped in building a well engineered product.

Integrated circuits are used for all the signal amplification. The power amplifier consists of a large high-power IC which is bolted onto a substantial heat-pipe which moves the heat away from the main circuit board and its heat-sensitive components. A single power supply is used.

£110 **MERIT** ★★★☆☆ **VALUE** ★★★☆☆

TEST RESULTS

Maximum power output was 50W with both channels driving 8ohms; the Akai really impressed me with its 4ohm capability, a figure of 100W being measured. The protection circuit cut in to limit the current when the output reached a respectable 13 volts RMS into the capacitive load. Up to that point the distortion remained low. Similarly good figures were measured for harmonic distortion, power bandwidth and damping factor, while the almost non-existent DC offset voltage was a credit to such an inexpensive amplifier.

Measurements made on the pickup input showed a reasonable level of performance although the hum level was a little on the high side. Against this must be set the good overload capability. The stereo separation figures were rather disappointing, being only about 25dB at 20kHz. The equalisation fell within rather wide limits being most in error in the 400Hz to 1kHz region.

On audition the tonal character of this Akai was of a slightly forward midrange and a slightly recessed top. At low levels the sound quality was pleasant enough but there wasn't much resolution of complex detail or revelation of harmonics, say on struck piano strings. The imaging was imprecise with the sound all piled up between the speakers. When the volume was increased the sound quality was almost unpleasant, the high frequencies becoming aggressively dominant and the bass getting out of control.

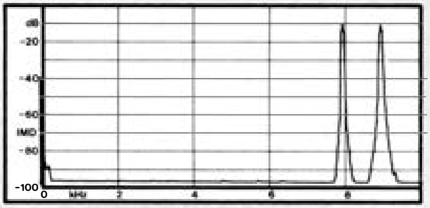
However, for it's price the AM-U33 did produce a sound that many people instantly recognise as hi-fi. Fed with undemanding source material the results would be acceptable to many owners

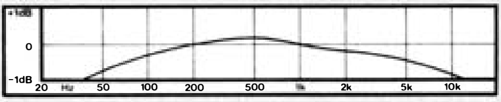
THE VERDICT

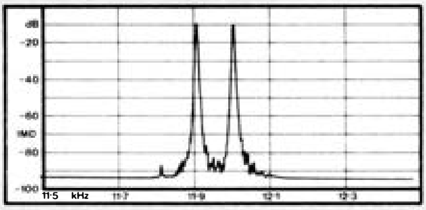
Although there must be some reservations over the Akai's sound quality, these were mainly concerned with the amplifier's reproduction at high levels. That said the AM-U33 offers a great deal for its price, particularly a generous power output capability and an overall above average performance. This amplifier is well built and nicely finished, so a recommendation will not be out of place, the overall rating representing good value.

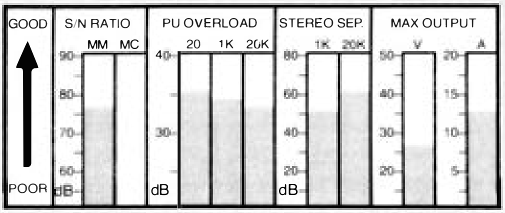
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	59	59	58
(WATTS)	8+8Ω	48	48	47
	4Ω	83	88	81
HARMONIC DISTORTION (% @ 35 WATTS)		0·015	0·008	0·009
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 17Hz		HIGH 17kHz









AUREX SB-66

The SB-66 is a 60W per channel amplifier and incorporates the Aurex 'Clean Drive' system which is intended to compensate for the signal losses that occur in the speaker cables.

This amplifier has a distinctive all- black appearance which is enhanced by the use of an acrylic fascia over the top section of the front panel. This fascia carries illuminated legends for the control switches and displays long coloured bars to signify their operation. The effect is quite striking and, for once, can be seen as a useful aid for checking the settings especially as the switches are very small and, having black buttons, difficult to see.

The SB-66 has 4 main inputs (both moving-magnet and moving-coil for disc) and two tape inputs with comprehensive dubbing (both directions) and monitoring facilities. The large diameter centrally placed volume control does not follow the fashion of detented rotation in steps but operates very smoothly with good tracking between channels.

The bass and treble tone controls cannot be bypassed but do have the centre positions detented. They are supplemented by subsonic and loudness filters. The control line-up is completed by a muting switch (−20dB) and a mono-stereo mode switch.

Two pairs of switched loudspeakers can be connected but only one pair has the 3-wire Clean Drive facility which can be turned-off by a front panel switch.

CONSTRUCTION & TECHNOLOGY

Virtually all the pre and power amplifier circuitry is carried on one very large PCB. Most of the push-button switches are soldered directly onto the board and then operated by long actuator rods to the front panel. The SB-66 has a common power-supply for both channels, which is fed from a substantial transformer.

Generally the components are of a good quality and the standard of manufacture high. Much use of ribbon cable keeps the wiring tidy although for some reason the transformer wires are not cut to length the surplus being bundled up and tied rather untidily.

The circuit design is simple and notable for the

£166 **MERIT** ★★★☆☆
VALUE ★★★★☆

minimal use of components. The disc stage uses an IC Op-Amp but with a discrete transistor differential front end to reduce the noise. The same circuit is used for moving-coil amplification by switching the input and feedback resistors (increasing the gain by a factor of 10). This is an increasingly common money-saving technique used by Japanese manufacturers but it hardly represents a design optimum and rarely sounds satisfactory.

The remainder of the preamplifier circuitry is based around an IC. The power amplifiers also use a simple circuit with a complementary output stage using low-cost plastic power transistors. Relay loudspeaker protection and the amplifier's limiter circuitry is controlled by an IC.

TEST RESULTS

The power-output of the SB-66 fell appreciably when it was hot after the pre-conditioning run so in normal use the 67W output (both channels 8ohms) would be exceeded by an appreciable margin. The power rose comfortably into 4ohms (110W) but fell back to about 80W into 2ohms because of protection limiting — still a quite acceptable result. The harmonic and IM distortions were low but an anomaly was shown in the damping factor measurements. This was about 60 at low and mid frequencies but had fallen to only 14 at 20kHz this representing an output impedance of nearly 0·6 ohms. This would result in a noticeable signal loss at high frequencies with some speaker types unless the Clean Drive system was in operation.

The measured performance of the preamplifier section was fine. The frequency response was fairly extended being down 1dB at 4Hz although the subsonic filter was very effective the response being down 2dB at 20Hz but 19dB at 6Hz.

The RIAA response was accurate within ±0·2dB limites 20Hz and proved to be virtually flat in character.

In the listening tests the use of the Clean Drive systems was found to give rise to only small audible changes and these varied with the choice of speaker. Generally though the lower mid-range became more detailed and less warm. Overall the sound quality of the SB-66 was not outstanding. The sound was pleasant and well balanced but the bass lacked real punch and there was a notable lack of resolution in the imaging so that the sense of front-to-back location of instruments was most vague.

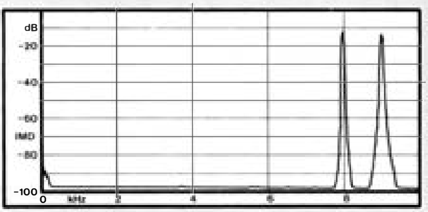
THE VERDICT

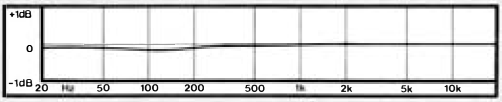
The Aurex SB-66 is an interesting looking amplifier having a quite commendable performance on the test-bench. It was thought to have an average sound quality, smooth, good tonal balance, but not too revealing or precise. Nonetheless it will be well liked by many who want a good all-round well-equipped amplifier. At its price the SB-66 must represent good value.

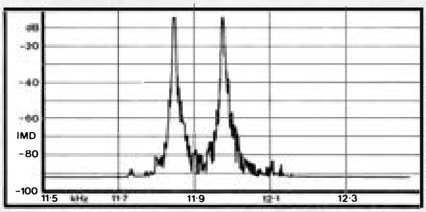
TEST RESULTS

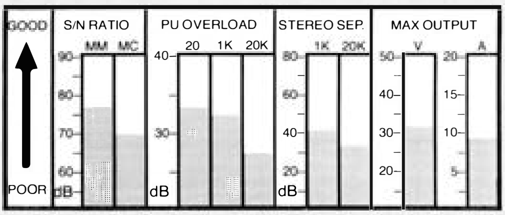
TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	75	79	79
(WATTS)	8+8Ω	60	67	65
	4Ω	102	110	108
HARMONIC DISTORTION (% @ 50 WATTS)		0·006	0·009	0·030

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	4Hz	50kHz









AUREX SB-77

The SB-77 is the big brother of Aurex's SB-66 Clean Drive amplifier and is rated at 100W in contrast to the latter's 60W. Less desirably the price has almost doubled from the SB-66's £165 to £330. Otherwise the two models are visually much the same and have many common features.

The styling of the SB-77 is quite distinctive, albeit rather flashy, with the all-black case setting-off an acrylic fascia occupying the top two-thirds of the front panel. This fascia carries the large volume control and a number of thin-bar indicators which light-up from behind; with a number of miniature (almost invisible) push-switches along its bottom edge. The remaining rotary controls are on the bottom third of the front panel.

There are four inputs (including both moving-magnet and moving coil for disc) and two tape inputs with comprehensive dubbing and monitoring facilities. The bass and treble tone controls can be switched out of circuit and are supplemented by loudness and subsonic filters. There is also a 20dB muting switch and a mode switch which also allows the stereo channels to be reversed. Two pairs of loudspeakers can be switched on together or separately, although only one pair can be connected in the 3-wire 'Clean Drive' configuration. On the back panel there is a switch which is normally locked into position which sets the amplifier for normal 8 or 4 ohms operation. For use in the UK the amplifier is supplied with the switch in the 8ohm position.

CONSTRUCTION & TECHNOLOGY

The chassis of the SB-77 is quite crowded with, in places, one PCB stacked above another and there is a fair amount of inter-board wiring fixed by wire-wrapping onto terminal posts and neatly located loomed into cableforms. Generally the standard of construction shows great care but the mains supply wiring and fuse assembly is untidy, although quite safe.

A large toroidal transformer is common to the both channels and the power supply uses generous sized 15000μF reservoir capacitors and such niceties as an extruded heatsink to keep the bridge rectifier cool.

No circuit details were provided but it was clear that a mixture of ICs and discrete transistors is used.

The 4/8ohm switch was found to change windings on the mains transformer in order to give a lower supply-rail voltage and, hence, a lower dissipation (and cooler running) by the power transistors, but equally a lower maximum available output.

The most interesting part of the SB-77's circuitry is the 'Clean Drive' system which uses a third wire to the loudspeaker terminals in order to sense the output current. This signal is applied to the input of the power amplifier as positive feedback and, causes the output voltage to increase by a factor great enough to compensate for the signal loss in the loudspeaker cables. The amount of positive feedback possible does not affect the overall stability of the amplifier.

£332 MERIT ★★★★☆ VALUE ★★★☆☆

TEST RESULTS

For the purpose of these measurements and tests the SB-77 was set to its 8ohm position. The both channel power output into the 8ohm load was about 120W rising to 140W with one channel only. There was also a healthy rise in output to 210W into 4ohms but the power into 2ohms was limited by the protection limiter as was the output into the reactive load.

The harmonic distortion was found to be very low right across the audio band and there were no signs of notable high frequency IM distortion.

The performance of the preamplifier sections was also found to be quite good. The noise levels of both moving-magnet and moving-coil inputs was fairly low and a good overload margin was achieved. The stereo separation of the disc input was quite fine at high frequencies but was rather low at 1kHz.

The RIAA accuracy fell between +0·2/−0·2dB limits 20Hz to 20kHz and the response curve showed a slight upper-bass, lower-mid lift and some slight treble roll-off.

The SB-77 was also tested in the 4ohm setting, the single channel power-outputs then showing a reduction to 80W into 8ohms and 130W into 4ohms. It seems obvious that this setting is to avoid any long-term overheating problems when driving 4ohm loads to comply with the German DIN tests.

For all the initial listening tests the SB-77 was operated into loudspeakers with all Clean Drive connections made and then a short appraisal was made using a pair of Monster cables. The sound quality of the SB-77 was more controlled than that of the smaller SB-66 particularly at the bottom end where the bass was quite firm and solid until fairly high levels. But the same 'depth' problem was apparent and instruments were not precisely located. It was also felt that to an extent there was a loss of resolution at high-frequencies and this resulted in a loss of delicacy during some musical passages.

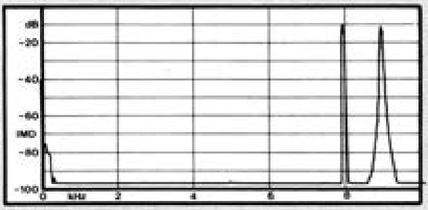
THE VERDICT

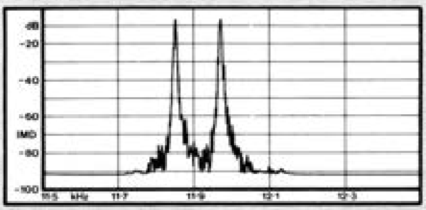
It was not found that the 'Clean Drive' system offered a significant enough increase in performance to justify more than a small price premium over its competitors.

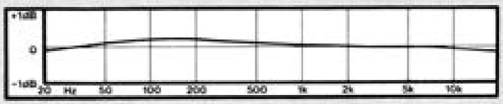
Therefore the SB-77, although a quite competently engineered 100W amplifier, may well be thought to be a little over-priced for the sound quality it offers.

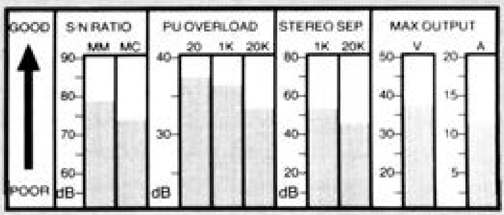
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	140	142	142
	8+8Ω	120	122	122
	4Ω	196	210	200
HARMONIC DISTORTION (% @ 100 WATTS)		0·010	0·003	0·009
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 2Hz		HIGH 46kHz









BEARD P500A P80

This combination is unusual in being the only valve amplifiers in this Buyer's Guide. They are both constructed with thick aluminium front panels, an all-over black paint job with white lettering, chunky, but rather crude, black knobs; the overall styling being like a big transistor amplifier in the American tradition.

The P500A preamplifier has two moving-magnet disc, two line and two tape inputs. Otherwise the design is 'straight line' with just balance and volume controls.

The P80 has undergone a major redesign of the circuitry since I last tested this 80W per channel model. However the external appearance remains the same with two large power (or more correctly output voltage) meters; a gain control per channel and a variable meter sensitivity control.

The P80 also has two alternative inputs which can be selected by a switch or the signal can be turned off.

The overall appearance of these two units was quite striking and gave an impression of power and solidity but the detail finish was rather untidy by Japanese standards.

CONSTRUCTION & TECHNOLOGY

The P80 has two mono amplifiers installed side-by-side in the same casing. Most of the components are mounted partly on PCBs and partly on tag-boards. There is a lot of hard wiring but this is nearly done with much use of clamps and ties to locate cableforms correctly. The chassis itself is very strong and rigid being made of heavy gauge steel.

The P500A is built into a similar steel case. Again the internal layout is very neat with the nine valves plugged into a large PCB which is stuffed with good quality components. The regulated power supplies are quite complex using two transformers and housed in a separate steel box within the chassis, no doubt in order to shield out mains hum radiated from those transformers.

No circuit information was provided although it was apparent that the amplifier stages used valves! Actually the P80 uses two KT88 valves in ultra-linear connection with fixed bias and large expensive-looking output transformers.

TEST RESULTS

Earlier this year I tested a P80 and obtained some disappointing result which led the designer to make some major chances to the circuit. These have obviously worked because the performance of this sample was considerably better. Power output into 8ohms was almost exactly 100W but fell slightly to 90W (less at 20kHz) into 4ohms because of the transformer mismatch. The peak instantaneous current was about average at 11 amps.

The harmonic distortion was perhaps higher than readers are used to seeing but was essentially low-order and as such relatively inoffensive. The IM distortion whilst low on the widely spaced tones increases to show some significant sidebands when the two tones 100Hz apart were used. This could result in some loss of resolution at higher frequencies.

The preamplifier measurements were quite acceptable although some might find the disc inputs rather too sensitive (approx. 1 mV gives 100W into 8ohms). The overall frequency response was sensibly curtailed at high frequencies but was well extended at the bottom end being barely 1dB down at 2Hz. The intention has apparently been to have minimum phase shift at low-frequencies. The RIAA response is essentially flat but shows a significant bass and treble roll-off — the former being quite surprising in view of the response through the aux. input.

The initial impressions during the listening tests were of a tight and solid sound with some midrange emphasis and a smooth almost rounded character.

The two units were auditioned separately and the P500A was thought to be the weaker of the products although the warm smooth quality will be popular with many listeners. Solo voice was particularly pleasant but more complex passages revealed a loss of some detail as well as a lack of precision in reproducing the 'attack' of some rock passages.

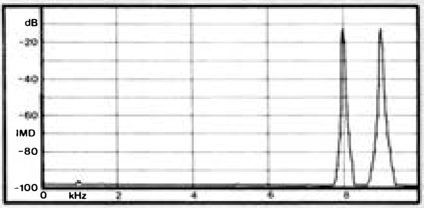
The P80, though, is now much improved with a firm bass reproduction that seems to drop the sound through the floor on loud passages! Yet the nice midrange quality has remained with a feeling of depth that was most convincing. The treble was felt to be slightly recessed and with some merging of detail.

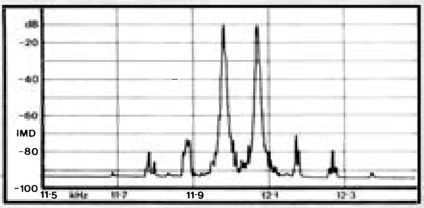
THE VERDICT

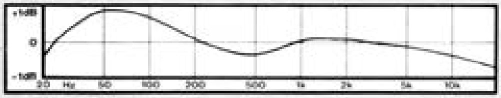
Obviously a product to polarise opinions and not one to appeal to the 'man on the Balham omnibus'. Yet there are many people who, once exposed to a good valve amplifier, can no longer come to terms with the sound made by the average transistor amplifier. This system has its own unique character and comes from a pretty well established manufacturer so I've no doubt that many readers will want to listen to this combination.

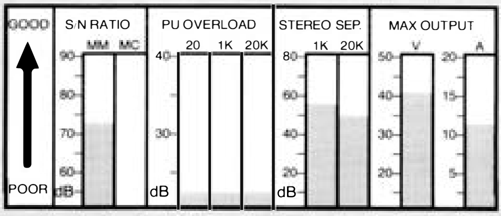
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	98	100	96
(WATTS)	8+8Ω	98	100	96
	4Ω	90	90	13
HARMONIC DISTORTION (% @ 80 WATTS)		0·270	0·106	0·602
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 2Hz		HIGH 35kHz







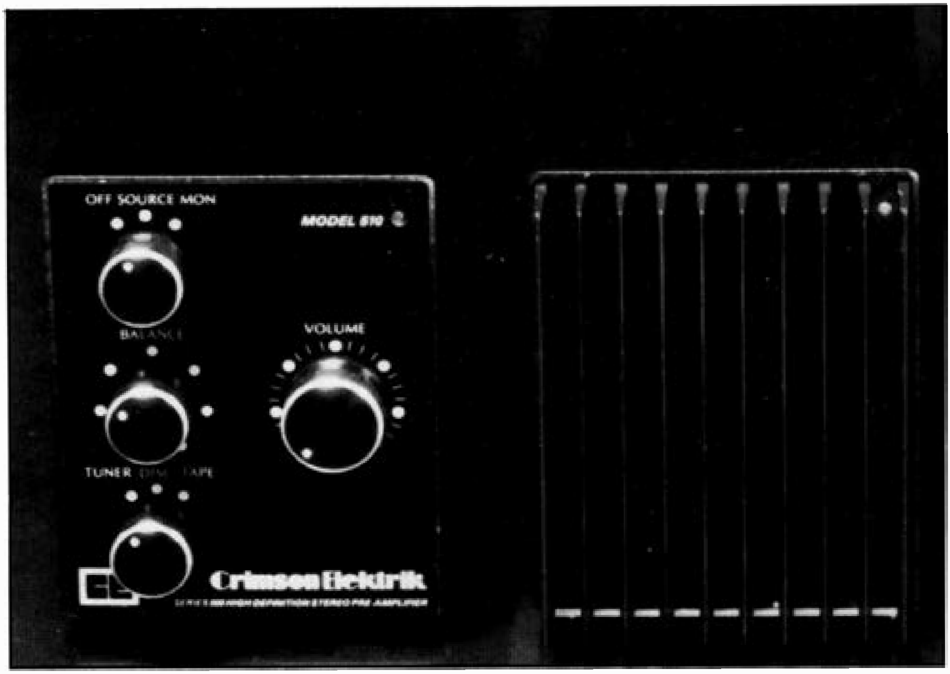


CRIMSON 510 530

Although they started in the business of making low cost kits but now the bulk of their activities is in producing a range of low-cost but highly individualistic audio amplifiers. The review models consisted of a 510 preamplifier fitted with a moving-coil input module (an alternative moving-magnet module is available) and a pair of 520 mono amplifiers. The styling or lack of styling gives the units something of a home-made appearance but the intention has been to spend the money on components that contribute to the final performance

The power amplifier has an LED indicator set into the heatsink but the input and output (4mm sockets) connections are at the back together with an awkwardly placed rotary on/off switch.

The preamplifier has a silvered panel with four fairly cheap looking knobs for volume, balance, tape monitoring and the input selector for the tuner, disc, and tape inputs. Inside there is space for a range of modules giving gain and loading options for different cartridges. But most of the space is taken up by the two PP9 batteries which are intended to give hum free operation. Batteries are expensive and also develop a high internal resistance as they discharge so a better long-term investment is a set of rechargeable nickel-cadmium cells and charger. One problem experienced was that we kept forgetting to turn off the preamplifier with the result that the batteries ran flat.



CONSTRUCTION & TECHNOLOGY

Internally the 530 could hardly be simpler with transformer and capacitors on the chassis and a power board bolted to the extruded heat sink. The layout is neat with tidy well-secured wiring but I was disappointed to find the capacitors secured by a tie-strap and sticky pads. The PCB is rather crowded but of good quality and printed with component designations. No circuit was provided but the 530 appears to be an AC feedback type with a quasi-complementary output stage using the well-regarded Toshiba 2N3773 transistors.

The 510 is also nicely built but in this case the bulk of the space is taken up by the batteries. Two main PCB's are used and these plug-in to each other; one being removable depending on whether the moving-magnet or moving-coil

£114 MERIT ★★★☆☆
£125 VALUE ★★★★☆

option is desired. The moving-coil stage circuit uses a simple two-transistor complementary stage but the remaining circuitry is moderately complex using a mix of discrete transistors and what appear to be BIFET ICS (although Crimson have rubbed off the type numbers)

TEST RESULTS

The maximum continuous average output power into 8ohms was 71W at 1kHz but was down to 58W at 20kHz this suggesting that a more accurate full-band power-rating would be 50W, not the manufacturer's 80W. The power rose nicely into 4ohms (100W) and to a lesser extent 2ohms (115W) but again the 20kHz figure was significantly lower because of rising distortion

The harmonic distortion was low in the midband but rose a great deal by 20kHz and some IM sidebands were revealed. The transformers used in the 530s also gave off severe audible mechanical hum during power-testing

The 510 preamplifier gave better results with fairly low-noise and an excellent overload figure However the stereo separation figures were poor at 1kHz. The RIAA equalisation was rather inaccurate by current standards and the curve shows a degree of bass cut becoming severe below 50Hz (−2·6dB at 20Hz) and a slight boost in the presence region. The same bass-cut was maintained through the auxiliary input the −1dB points being 24Hz and 30kHz.

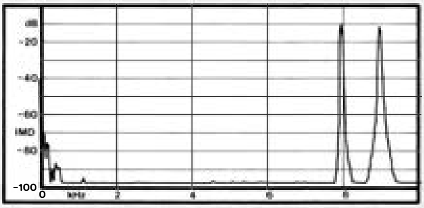
The results of the listening tests were rather more positive, the sound stage being quite stable with good imaging and, to a degree, controlled delivery of power. The bass was solid in character and had good definition — indeed it would seem that Crimson have made a worthwhile exchange of low bass extension for bass clarity. The midrange was open but rather forward and this gave a good sensation of action and dynamics on rock programme material. The only real problem area was at the top end which was tending to hardness and coarseness. Otherwise a good result.

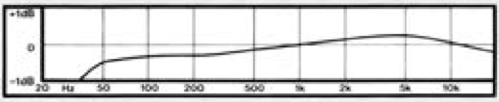
THE VERDICT

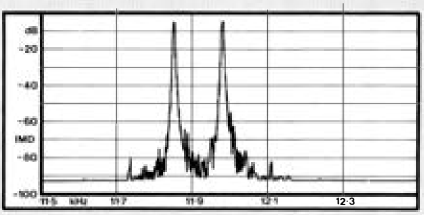
Nothwithstanding concern about the treble qualities this combination was pleasant to listen to. Its appearance is in the best cottage-industry tradition although the internal construction was fine. On a value for money basis the 510 preamplifier must be a bargain and our tests showed that it worked well with other power amplifiers. As we went to press Crimson arrived with new, all-black, versions of the 510 and 530. This version of the 530 delivered some 90W into 8ohms at 1kHz. Otherwise there was no sonic difference between the old and new models. So the 530 power amplifier can be seen to represent reasonable value, but it does face some severe competition.

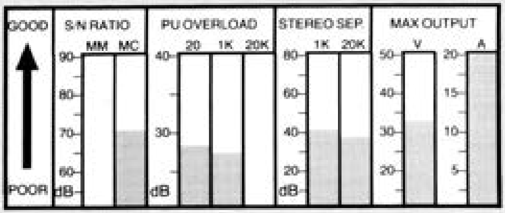
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	87	90	75
(WATTS)	8+8Ω	87	90	75
	4Ω	130	139	92
HARMONIC DISTORTION (% @ 70 WATTS)		0·008	0·004	0·017
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 24Hz		HIGH 30kHz

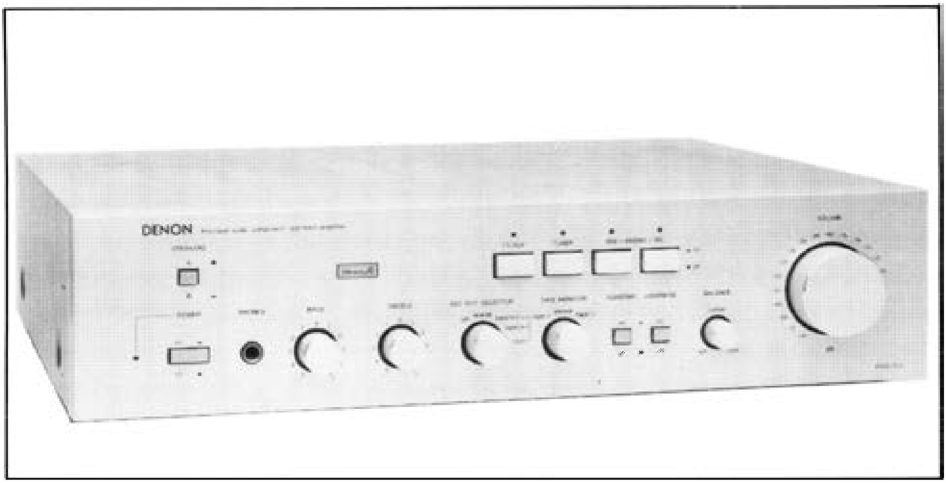








DENON PMA-510



In many ways the Denon PMA-510 looks the archetypal Japanese amplifier with its brushed aluminium fascia and slimline appearance. Rated by the manufacturer at 50 watts per channel the PMA-510 has four main inputs (selected by the four clearly positioned push-buttons) including both moving-coil and moving-magnet cartridges. Two tape-decks are provided for, the 'Rec. Out' switch enabling the user to record the signal from any input source (not necessarily the one being listened to) and to dub from deck to deck in either direction. The tone-controls cannot be bypassed to ensure a 'flat' setting but they do have centre detents and these were, in fact, found to be accurate. The usual loudness control is provided, together with a subsonic filter turning over at nominally 20Hz, although with its slow 6dB/octave slope it proved to be of little practical use.

Two pairs of loudspeakers can be connected but the single switch only allows one pair to be connected at once. Perhaps one of the better features of this model is the dominant volume control which fitted the hand nicely and proved to have a smooth progressive action with accurate tracking between channels over most of its range.

In all, a neat unadventurous design finished to the excellent high standards we have come to expect from Denon even in a product of such low price.

CONSTRUCTION & TECHNOLOGY

My first impression after studying the internal construction of the PM-510 was to wonder how Denon managed to manufacture it for the price. That fine front panel is not an aluminium extrusion or a plastic moulding but a precision diecasting. The same philosophy extends to the choice of components with a high quality volume control, dust-sealed (instead of open) preset pots, and an oversize extruded heatsink extending the full amplifier width, doubling as a chassis support to improve rigidity. The circuitry is on two boards, the rear one of which is connected to the small signal sockets. All the signal switching is done on this board using remote rod and cable operated switches. The front board carries the tone controls, etc, the two power amplifiers and the single shared power supply which is fed from an unusually large transformer.

One nice piece of attention to detail is the way in which all cables are firmly located by clamps to the chassis, thereby avoiding any likelihood of the performance changing if the cables are accidentally repositioned.

No circuit information was provided by the importer but an inspection revealed that all the preamplifier circuits are based around the use of low noise ICs made by NEC. The power amplifiers use discrete transistors in a DC-coupled circuit with a complementary output stage. Two biasing networks are used, one of which maintains a bias on whichever output transistor would be cut-off by the signal. This gives a non-switching configuration free of crossover and notch-type distortion.

£139 MERIT ★★★★☆ VALUE ★★★★★

TEST RESULTS

The PM-510 proved to have a healthy output capability with some 70W available on single channel operation into 8ohms rising to an excellent 110W into 4ohms. There was hardly any difference between the single and both channel power-output, a testament to the generous power-supply fitted. Harmonic and IM distortion both proved to be very low with, as expected, no visible crossover artifacts. The peak current capability was determined by the saturation of the output transistors, rather than by protection limiters, and was an adequate 17 amps.

The stereo separation was also excellent with 61dB at 20kHz. Relative to the measured sensitivity the pickup overload margins and the noise figures were also good. The overall bandwidth was sensibly limited outside the audio band whilst the RIAA equalised response was very accurate.

The Denon PM-510 gave a performance during the listening tests which was surprisingly good for an amplifier of this price. When used at sensible levels it was hard to find much wrong with this model except by making unfair comparison to hyper-expensive exotica. The sound was clear, with good tonal balance and accurate resolution of stereo images. The sound was also detailed, although the bass could become a little muddled on full orchestral passages. Unfortunately the sound became progressively more unpleasant as the volume was raised with a sort of 'reedy' character dominating.

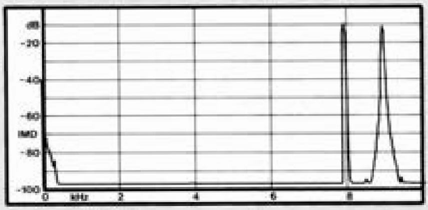
THE VERDICT

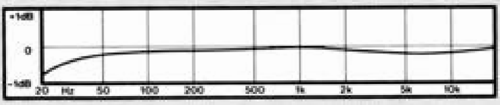
The PM-510 is a well thought-out product with a useful range of sensible rather than gimmicky facilities. Construction, finish and, to an extent, styling are all to Denon's usual high standard. The laboratory measurements showed an above-average level of performance while the listening audition found little to criticise and a lot to praise provided that the listening level is moderate.

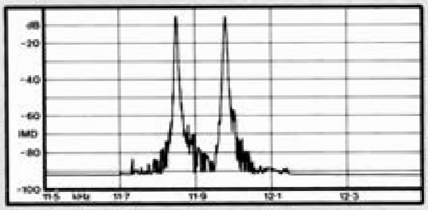
In all a product well worth its recommendation at this price level.

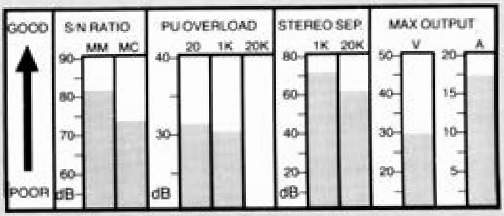
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	69	70	70
	8+8Ω	62	68	66
	4Ω	106	110	106
HARMONIC DISTORTION (% @ 50 WATTS)		0·019	0·013	0·013
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 10Hz		HIGH 48kHz









DENON PMA-540

The PMA-540 is obviously similar to the PMA-510 model, although the power output is increased to 80W and a few additional features are provided. The standard of finish is high with all the controls fitting accurately and working smoothly. The silver-grey finish gives the fascia a soft, attractive appearance although it does seem rather cluttered compared with the simpler PMA-510.

The four inputs (including one for moving-coil cartridges) are selected by the row of rectangular switches, each one of which lights up an indicator. A further selector switch permits dubs to be made in either direction from the two tape connections or it can switch off the signal feed to the recorders. Other controls include loudness, an inadequate 6dB/octave subsonic-filter and a 'direct' switch which bypasses the bass and treble controls. The Denon can drive two pairs of loudspeakers and these can be individually switched.

Denon make much of their 'Real Time Audio Technology' in the 540. This philosophy covers several different techniques including direct-coupling and DC servo systems to maintain absolute stability; also use of a non-switching type of output stage — another one of the pseudo Class A circuits, this one being called 'zero cross linear bias'.

CONSTRUCTION & TECHNOLOGY

Like the less expensive PMA-510 this Denon model is constructed to a very high standard, typified by the use of a diecast front panel finished to a very high standard. Inside, the layout of the PMA-540 is neat with just two circuit boards interlinked by cables and wires firmly retained in position by clamps attached to the chassis. Nonetheless, the main circuit board looks untidy with a lot of wires criss-crossing its surface.

A large extruded heat-sink extends across the middle of the chassis and in consequence space is short so the large conventional transformer of the 510 has been replaced by a smaller (but presumably more efficient) toroidal type. Again a single power-supply is shared between the channels.

The circuit design seems similar to that used in the PMA-510 and again uses ICs throughout the preamplifier stages. The moving-coil stage, though, appears to be enclosed in a metal screening can, no doubt to reduce the pickup of mains hum and interference spikes.

The transformer is not double-insulated so this amplifier must be used with the earth wire connected. I was also surprised to find that only the live side of the supply is switched (not neutral as well) and that there are no visible secondary fuses — only a single 8 amp mains fuse.

TEST RESULTS

Despite its apparent similarity to the 510 the PMA-540 is a more powerful design but doesn't have a proportionally larger power supply. In consequence there is some difference between the single and both-channel power outputs although both figures were comfortably over the rated figure. The power rose into 4ohms, but not as much as expected, there being indications that the protection circuits were beginning to operate. Into 2ohms the power fell as the protection clamped the output. The biggest shock was the peak current capability which was substantially below that of the PMA-510 (17 amps) — not much more than 6 amps being possible for any longer than a very short transient.

Both harmonic and IM distortion were very low and unlikely to be of any significance.

The preamplifier stages measured quite well with good stereo separation right across the audio band and an adequate overload margin, although this fell slightly at high frequencies. The disc inputs were surprisingly noisy and there was some audible hum which we were unable to eliminate. Unlike the PMA-510 the 540 has an extended bandwidth (unnecessarily extended I believe) with a −3dB response from 2Hz to 240kHz.

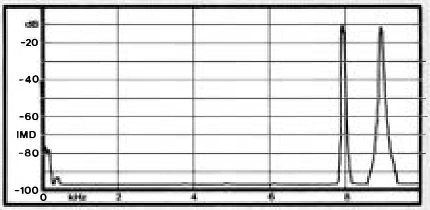
After auditioning the PMA-510 I expected great things of the 540, but unfortunately it did not live up to the promise. At low powers the sound was pleasant but undistinguished, but at moderate and high levels the quality was degraded, the bass in particular being loose and uncontrolled with, in consequence, little resolution of low-frequency detail. On occasion the high frequencies seemed 'ragged' and as a result the amplifier could not be played at the sort of levels the power ratings would suggest.

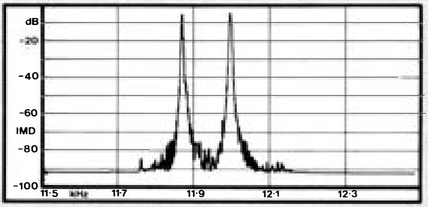
THE VERDICT

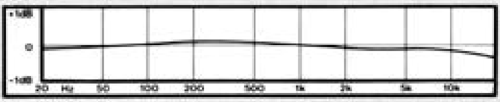
The Denon PMA-540 has all the right ingredients yet, despite its wide range of facilities and superb standard of finish, it only achieved an average measured performance and below average sound quality. In fact the less expensive PMA-510 represents a better buy and definitely a more pleasant listening experience. It should be easy for Denon to pinpoint the differences and improve the 540 but in the meantime it does not figure as one of our recommendations.

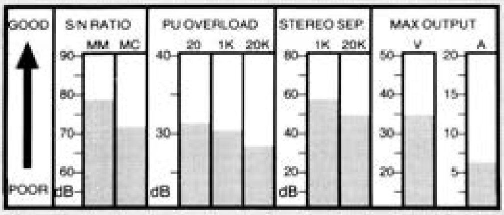
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	110	115	112
(WATTS)	8+8Ω	94	100	90
	4Ω	145	145	156
HARMONIC DISTORTION (% @ 80 WATTS)		0·005	0·005	0·005
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 3Hz		HIGH 50kHz







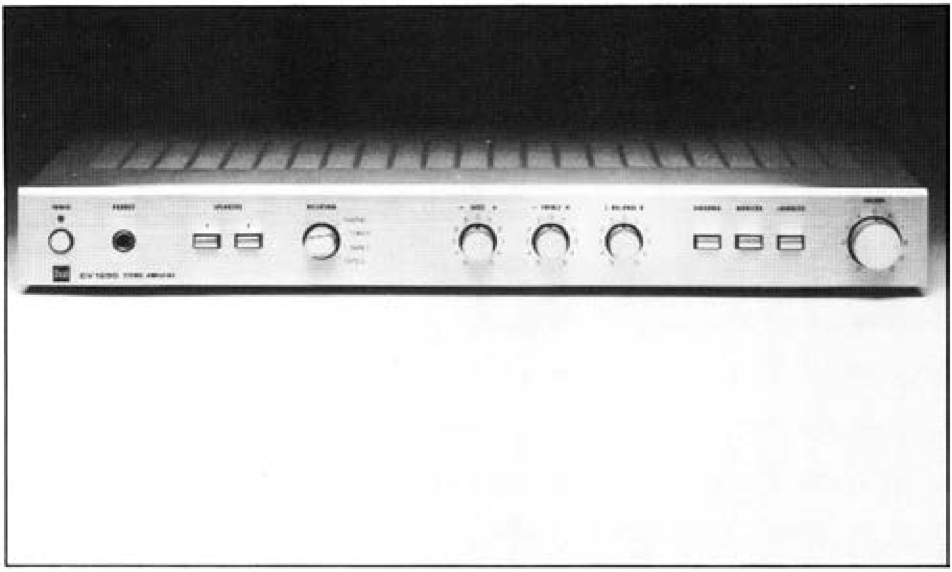


DUAL CV1250

This West German product is one of Dual's more recent designs. Its styling is notably European, with the long low look and an anodised front panel on a silver-grey case. The front panel layout can be confusing initially for those reared on generations of Japanese amplifiers but it is logical in a funny sort of way. There are a few normal inputs plus not two but three tape sources. Tapes 1 and 2 can only be selected by the source function switch whilst Tape 3 (labelled as Monitor) can only be heard by, yes you've guessed, pressing the tape monitor switch. Some more of that funny logic. Two push-buttons select loudeness and subsonic filter with a quite effective 18dB/octave slope and turning over at 18Hz. The bass and treble tone controls cannot be switched out of circuit but do have detents in the zero position. Next to the headphone socket there are switches for the two pairs of loudspeakers.

On the back panel there is a slide to give a choice of sensitivity on the disc input (nominally 1·5mV and 5mV for rated output).

Unlike most of its Japanese competition the Dual is bare of meters and flashing lights, there just being a single LED for power on.



CONSTRUCTION & TECHNOLOGY

The chassis layout is rather untidy but the standard of workmanship is high and good quality components have been used. In particular a large extruded aluminium heatsink is fitted and, just to be sure, several over-temperature safety cutouts.

The circuit design of the CV1250 is unusual for a budget priced amplifier. All the input signals are switched electronically using a CMOS IC (the same technique is used in the Quad 44 preamplifier). The use of these ICs has enabled Dual to switch the signal at the input sockets and so avoid the need for signal path wiring and the risk of noise and crosstalk pickup along the cable path. Before the switches only the disc signal is amplified this being done by a low-noise IC.

After the volume control there is another IC and the active subsonic filter followed by a passive tone-control network. This is very unusual these days because this passive network will intrduce about 30dB of signal loss.

The power amplifer is one of the increasingly familiar power IC packs, both channels being driven from a rather small power supply.

£105 MERIT ★★☆☆☆ VALUE ★★★☆☆

TEST RESULTS

The measurements of continuous average output showed that a 30W rating would be about right. Some 49W was available with one channel driving 8ohms but this fell to 40W with both channels loaded. The power into 4ohms rose but not as much as might be expected. This didn't seem to be the result of any limiting action but because the small power supply had high regulation and therefore collapsed when high currents were drawn.

The harmonic distortion was very low and the IM sidebands were of low-order. The frequency response was curtailed quite sharply outside the audio band, the −1dB points being 11Hz and 24kHz.

The disc stage had good stereo separation and overload margins but the noise level was a little high and, as can be seen from the plot, the RIAA equalisation showed a slight treble-boost and bass-cut. But the accuracy of this response is very dependent on the position of the tone-controls; in the zero position there was a more pronounced bass cut.

The subsonic filter worked well, the response remaining virtually flat to 24Hz but being 12dB down at 12Hz.

And so to the listening tests where first impressions suggested a not unpleasant but equally a not exceptional sound quality.

Overall the tonal quality was rather 'middy' which gave a false impression of detail although in fact there was some masking of low-level detail. The bass was well controlled and firm but this couldn't be sustained at higher levels where it lost any sense of power or 'punch'.

The treble was equally well controlled but lacked detail and gave distinctly vague imaging. A sense of depth (essential to realism) was also missing.

THE VERDICT

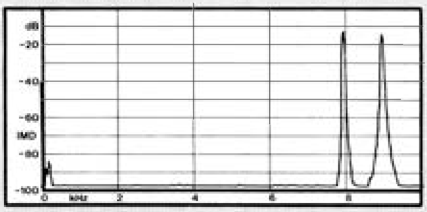
Despite a sound quality that was neither neutral nor accurate the Dual left a better impression than most amplifiers of its price. It proved quite capable of driving the more difficult loads which leave many competitive products gasping and turned in a measured performance showing few design weaknesses.

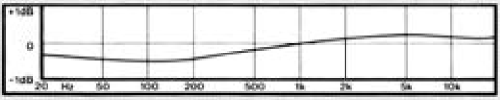
If you want a low-cost hi-fi amplifier and don't mind rather conservative styling, this Dual product could merit consideration.

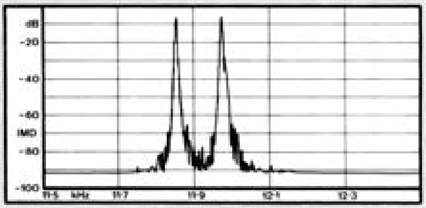
TEST RESULTS

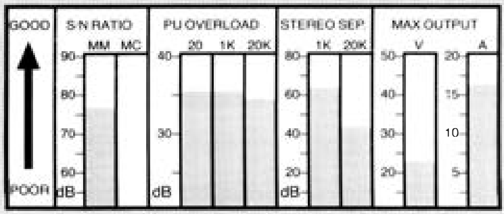
TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	48	49	49
	8+8Ω	40	40	40
	4Ω	60	66	64
HARMONIC DISTORTION (% @ 30 WATTS)		0·015	0·005	0·090

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	11Hz	24kHz









GRUNDIG SXV6000 A5000

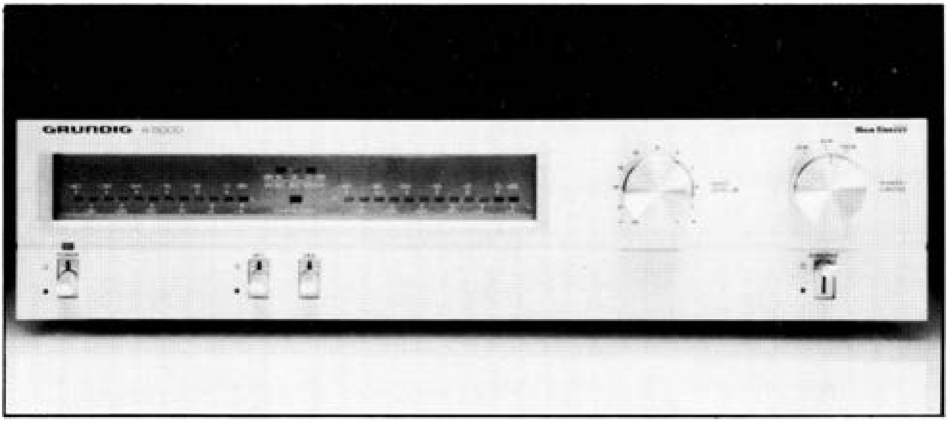
In recent years we haven't automatically associated the name of the electrical giant Grundig with hi-fi so it was interesting to see this pre/power combination from the German company.

The appearance of the preamplifier is a little gaudy with silver knobs on a shiny silver-grey fascia which, is covered by an array of knobs, indicators and tiny pushbuttons. The three large rotary controls are for volume, balance, and level but this is a misnomer for in fact it seems to be a variable loudness control to vary the normally fixed frequency contour at different listening levels. There are two normal inputs, tuner and disc (moving-magnet or moving-coil) and two tape inputs with cross dubbing which is poorly explained in the manual and none too clear from the front panel legends. The tone control section is quite comprehensive with the normal bass (40Hz) and treble (16kHz) controls being supplemented by two further cut/boost controls centred at 300Hz and 2500Hz. All can be defeated by the tone switch. Finally there is an internal 400Hz tone generator which can be used for setting-up the recording levels of tape decks etc.

The connectors on the back panel are an interesting mix of phono, 5 pin DIN, and 7 pin DIN types so a selection of adaptor cables may be advisable.

The A5000 power amplifier is probably best rated at 70W into 8ohms as DIN power specifications can be made to prove almost anything. It has a rather superfluous level control which changes the gain in accurate 3dB steps and a decent 18dB/octave subsonic filter. There are two switched speaker connections and naturally a great big 'power-meter' display.

Finally an interesting power-limiter system for those who are neurotic about their small speakers. Its labelled for maximum outputs of 30, 60, and 120W (4ohm) but is somewhat unnecessary except perhaps at parties when enthusiasm with the volume control might overcome prudence.



CONSTRUCTION & TECHNOLOGY

Only one word adequately sums up the constructional standard reached by Grundig — excellent! The chassis layouts were a joy to behold. In the preamplifier the circuitry is on two large, neatly laid out PCBs. The moving-coil circuitry is completely screened (for low-noise) by a metal box and further screening surrounds the large high quality C-core transformer which carries an over-temperature thermal cutout switch.

The power amplifiers each have a massive cast aluminium heat sink and separate power-supplies (rectifier and reservoir capacitors) but a common and massive C-core mains transformer.

The circuitry of these Grundig products is all discrete with some quite novel design work. Separate amplifiers are used for the moving-magnet and moving-coil inputs, the former using a differential input stage; the latter a single ended input. All the stages are powered from a single 30 volt supply rail although this is increased to 50 volts for the output amplifier stage

The fairly conventional power amplifier is preceded by an active subsonic filter which has 3 poles to achieve a sharp 18dB/octave roll-off.

£188 MERIT ★★★☆☆
£188 VALUE ★★★★☆

TEST RESULTS

The Grundig power amplifier was good for about 80W into 8ohms with little difference between the single and both channel figures; a testament to the efficiency of the large transformer used. The power into 4ohms rose to a very good 150W (midband) but the protection limiter was just coming into operation causing the 2ohm power to fall back to 70W.

The distortion was fine in the midband but rose sharply at high-frequencies and some IM side-bands were revealed although they were low in amplitude.

The disc amplifier was fairly quiet and the tone controls were found to have nicely symmetrical and well spaced responses. The overall frequency response was very tightly limited to the audio band; the −1dB points being 10Hz and 26kHz. The RIAA response was tolerably accurate and again showed a roll-off at either extreme — probably a good idea to help the power amplifier maintain control of the signal.

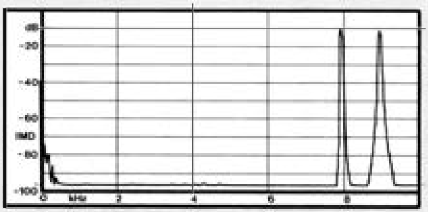
The results of the listening tests suggested that the Grundig was slightly below average in terms of sound quality. The performance of the power amplifier was quite competent with a well-controlled bass being noted but also a slight treble harshness and poor resolution. The preamplifier was the weak link for the sound quality, while clear, was lacking in detail and gave poor overall imaging. In many ways though this combination sounded the way most people expect hi-fi to sound.

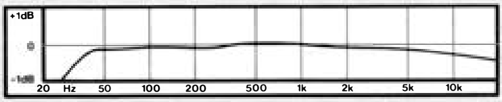
THE VERDICT

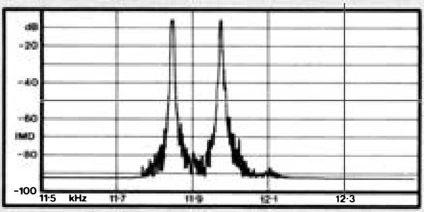
For the power and facilities offered the combined price of around £370 is very fair and many readers may see it as a good buy. In typical German fashion the production engineering and construction were both clinically exact and deserve praise while in the laboratory the meters gave the right readings. Yet even Karajan recordings revealed sonic weaknesses important enough to preclude an outright recommendation although the Grundig is not a bad amplifier combination.

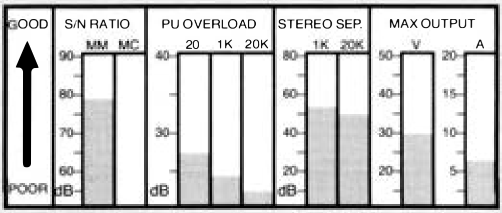
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	82	84	83
(WATTS)	8+8Ω	76	80	80
	4Ω	132	150	141
HARMONIC DISTORTION (% @ 70 WATTS)		0·027	0·023	0·170
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 10Hz		HIGH 26kHz









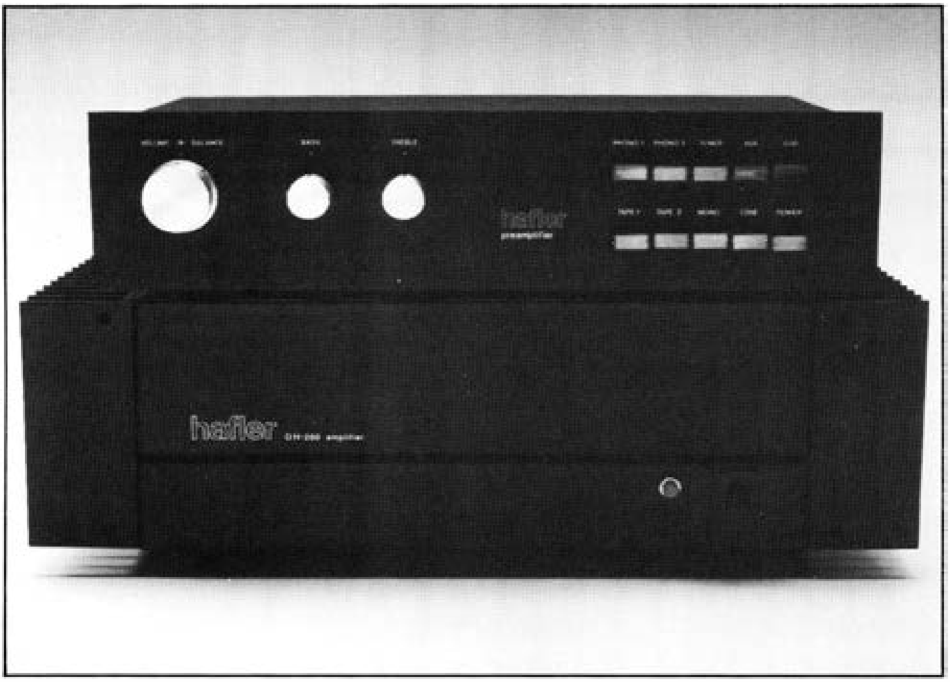
HAFLER DH101 DH200

This Hafler pre/power combination actually consists of three items: the DH101 preamplifier; the DH102 moving-coil amplifier (bolted inside the DH101); and the DH200 power amplifier. These products are available ready-built, or at a lower price, as kits. Their styling is pretty basic, both units being built into fairly crudely finished black boxes. Similarly the controls and switches are functional as distinct from elegant.

There are four inputs of which two are for disc so with the DH102 module one could be made moving-coil, the other remaining moving-magnet. There are also two tape connections but only simple dubbing is possible.

The remaining push buttons are for mono mode, tone-cancel and power. The rotary controls consist of a concentric balance and volume controls and separate bass and treble tone controls.

The DH200 is a 100W power amplifier using MOSFET output devices. It has one pair of unswitched speaker connections and no controls other than the front panel power switch next to the indicator lamp.



CONSTRUCTION & TECHNOLOGY

The review samples apparently started life as a kit and passed through a few hands before mine so it wouldn't be fair to comment on the standards of workmanship without first inspecting a factory-built sample. However, both units were apparently checked by the importers before being submitted to our laboratory tests.

The chassis layout of both units is neat and well organised although there is rather a lot of point-to-point wiring which can look untidy.

The power amplifier uses a common power supply for both channels. The output devices are Hitachi MOSFETS and two parallel pairs are used in each channel to increase the output current capability. The circuitry is mirror-imaged throughout the power amplifier with complementary long-tail pairs at the input followed by a Darlington voltage-amplifier and the driver transistors. Very simple, straightforward and well-conceived.

No circuit was provided with the preamplifier but investigation revealed that the disc amplifier also uses a mirror-imaged circuit, in this case a two stage complementary arrangement with the RIAA network in the feedback loop.

The second stage after the selector switches and volume control uses two differential pairs followed by a push-pull output stage and has the tone-controls in the feedback loop. IC type regulators are used to give a low-noise ±18 volt.

TEST RESULTS

The Hafler proved to be a powerful amplifier with some 130W output into 8ohms (both channels driven). This increased to 225W into 4ohms and 290W into 2ohms (for short periods before the fuses blow) the limits being set by the power supply rather than the protection circuits.

The harmonic and IM distortions were quite low; the slew-rate fine at greater than 5; and the damping factor above 100 at all audio frequencies.

The disc amplifier could have been quieter and did prove rather sensitive to sources of external mains hum. A fine disc overload figure was recorded and the stereo separation was adequate but not good.

The overall frequency response was sensibly curtailed outside the audio band and the RIAA equalisation, whilst accurate above 1kHz, was rather inaccurate below and showed some bass roll-off (see curve).

The Hafler did very well in the listening audition. It proved to be particularly capable of reproducing the sense of dynamics in the music; the sound also being clear and open with much detail being preserved. On the Quad speakers the imaging was superb with an accurate and stable sound stage outstanding in its front-to-back detail. But as is invariably the case, the Hafler did have a weakness the tonal response seeming to be rather too bright — enough to be mildly irritating after a while (although the use of the Studio One speakers cancelled out this problem).

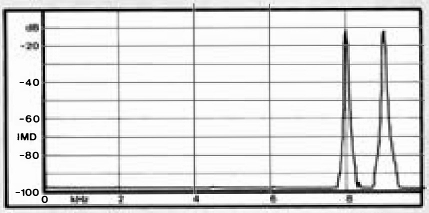
The moving-coil stage also worked well and was, subjectively at least, one of the quietest in the book.

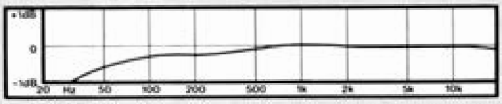
THE VERDICT

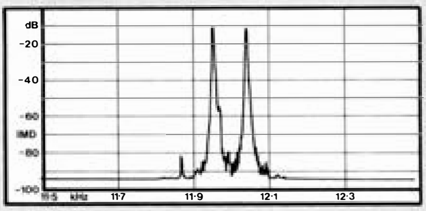
A moderately expensive combination (although you can save money by buying it in kit form) but one which performed very well. As with so much imported US equipment the transformer was prone to mechanical hum, but this proved to be a tolerable fault. The sound quality achieved was very good and despite the excessive brightness it will be popular with those who like a solid, well controlled sound with a seemingly effortless power-delivery and a great sense of being there. Recommended for further listening.

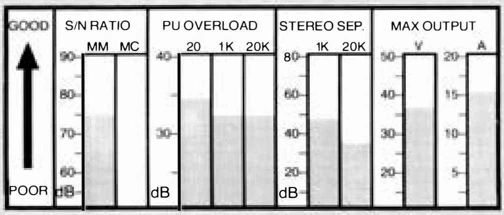
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	148	148	158
(WATTS)	8+8Ω	130	135	135
	4Ω	225	225	225
HARMONIC DISTORTION (% @ 100 WATTS)		0·021	0·007	0·020
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 7Hz		HIGH 32kHz







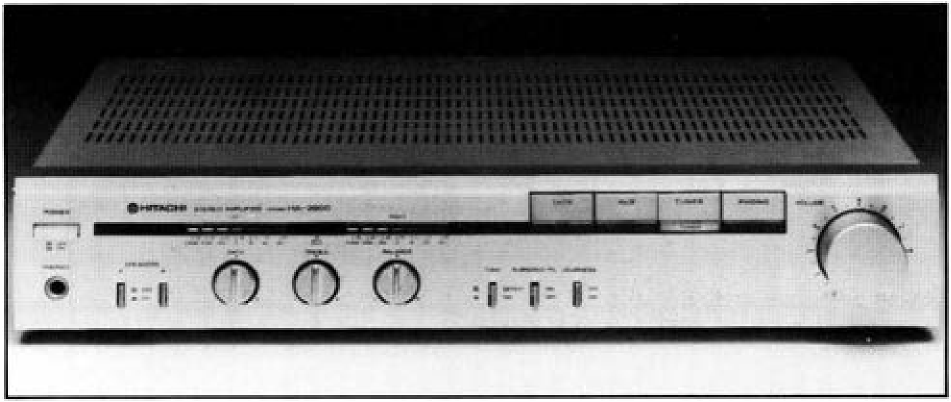


HITACHI HA-3800

The HA-3800 is one of the less expensive of Hitachi's integrated amplifiers, this one being rated at 35 watts per channel. It has a neat, almost slimline appearance, the silver grey fascia actually being a high-quality plastic moulding. Only the basic facilities are offered with three inputs (disc moving-magnet) and one tape connection. The bass and treble tone-controls can be bypassed by a push switch while other buttons operate a subsonic filter, a loudness filter and the two pairs of loudspeakers.

The Japanese majors believe, no doubt with some commercial justification, that their amplifiers must have power level indicators and the HA-3800 has two horizontal rows of seven LEDs neatly located above the tone controls.

Although this is a simple uncomplicated amplifier the standard of finish and little touches of quality were quite impressive.



CONSTRUCTION & TECHNOLOGY

The construction of the 3800 is both neat and simple. A single PCB carries what little circuitry there is and it is bolted to a small extruded heat sink. Despite the obvious emphasis on low manufacturing cost the quality of the components is high.

The circuitry consists entirely of a few IC's with two in the preamplifier and a large Sanken power IC comprising the bulk of the power amplifier design. This IC contains over-current and over-temperature protection but as an insurance against its failure there is a fuse in each speaker line.

£99 **MERIT ★ ★ ☆ ☆ ☆** **VALUE ★ ★ ☆ ☆ ☆**

TEST RESULTS

The rated power-output was exceeded by a small margin but the output into 4 ohms was limited to 25 watts because of this amplifier's limited output current capability (only 5 amps peak was measured). When loaded with 2 ohms little over 20 watts could be achieved. So obviously the choice of loudspeaker could be critical with this amplifier.

The harmonic distortion was fairly low in the midband but rose to 0·1% at 20kHz. Some sidebands were revealed by the IM tests but they were quite low in level. Although the high frequency was sensibly rolled-off at around 40kHz the low-frequencies fell away far too early (with the tone-controls 'flat') and some bass boost was necessary to achieve a more nearly-flat response. After such minor 'tweaks' the RIAA response was at its flattest but still showed bass and treble cut together with some lower midrange boost.

The other measurements revealed a highish noise level and poor stereo separation at high frequencies. The disc overload was quite adequate at 32dB.

The listening tests revealed that this Hitachi did not maintain good control over the sound except at very low-levels. The sound also lacked a sense of dynamics — a rather stodgy atmosphere being noted! However at low listening levels the sound quality was found to be clear and detailed; in fact surprisingly good.

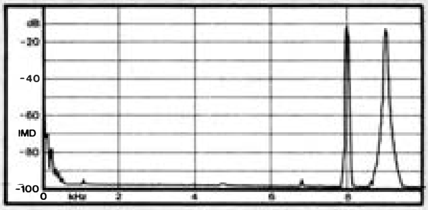
Tried with Roger's LS3/5As the combination proved pleasantly successful so it would seem that most of this amplifier's ills come from its inability to correctly drive many loudspeakers.

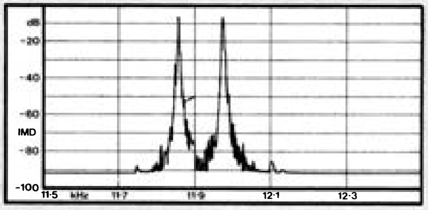
THE VERDICT

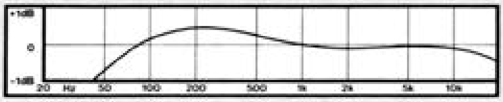
A smartly finished amplifier at a sensible price and one that could appeal to a lot of people. However it only has a limited output capability and so considerable care needs to be taken if an acceptable sound quality is to be achieved. In many ways the HA-3800 is classic evidence that if more money is spent on finish, appearance, and facilities; then less will be spent on the circuitry.

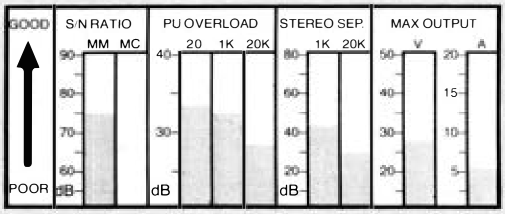
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	54	58	56
(WATTS)	8+8Ω	40	46	45
	4Ω	44	46	43
HARMONIC DISTORTION (% @ 35 WATTS)		0·032	0·022	0·110
FREQUENCY RESPONSE BETWEEN –1dB POINTS		LOW 58Hz		HIGH 38kHz









HITACHI HA-6800

The HA-6800 is a powerful (70W per channel) rather aggressively styled integrated amplifier finished with an attractive greyish-black painted colour scheme. The front panel is visually split, the top half carrying the power and input selector buttons and the large diameter volume control. Three normal inputs (disc can be switched moving-magnet or moving-coil) are provided and their selection is confirmed by small LED indicators. There are also two tapes deck connections with comprehensive dubbing and monitoring facilities.

On the lower half of the panel there are bass and treble tone-controls, which can be totally bypassed by pressing the tone-cancel switch. There is also a half-hearted subsonic filter (6dB/octave) and the perenial loudness filter. Finally this amplifier has a stereo/mono mode button and connections for two pairs of speakers and a rotary switch offering separate and both together operation

CONSTRUCTION & TECHNOLOGY

The chassis of the 6800 is pretty crowded and dominated by the very large dual heat pipe which is used to cool the MOSFET output transistors. It also acts as a shield for the disc amplifier board which is mounted along the right-hand side of the case although there might be some worries about the effect of the potential temperature rises on this sensitive circuitry.

The bulk of the circuitry is on a group of four neatly laid out PCB's linked by cable forms. Good quality components are used throughout with much use of fusible resistors (they fuse open circuit if they are overloaded) to backup the supply fuses in the event of a fault and so minimise the damage. A common transformer and power supply is used for both channels.

The preamplifier circuitry is almost completely made up of Hitachi's own ICs. The disc amplifier uses a long-tail pair of low-noise transistors ahead of the well-known HA12017 IC. Matching to moving-coil is achieved by the low-cost solution of switching in a 100ohm input resistor and a parallel feedback resistor to increase the stage gain by about 17 times. The tone-control amplifier uses another HA12017 IC for each channel but the power amplifier is entirely discrete.

The input differential pair is a dual FET with cascode transistors and is followed by a second differential pair which drive the complementary pair of output MOSFETS. The power amplifier also has a DC servo circuit to ensure good DC stability. The output signal is filtered to remove all the AC signal and the residual DC is amplified and used as a negative feedback error signal.

£199 MERIT ★★★☆☆ VALUE ★★★☆☆

TEST RESULTS

The power output into 8ohms was high with little difference between single and both channel operation. However the increase of power into the 4ohm load was limited by the protection so only 126W could be achieved instead of the expected 180W or so. The power into 2ohms fell back to 65W for although the peak current capability was 11amps this could only be sustained for the briefest periods falling back to about 8amps thereafter.

The harmonic distortion was low in the midband but rose at high frequencies. This non-linearity although small was also highlighted by the IM tests which revealed a number of sidebands down in the −70 and −80dB region

The performance of the preamplifier stages was quite adequate with few complaints to be made. The RIAA accuracy was very good showing a virtual ruler flatness from 20Hz to beyond 20kHz.

This Hitachi gained above-average marks in the listening tests for its clear, open and reasonably detailed sound quality. The stereo imaging was fine but the front-to-back depth was rather too 'flat'. The bass was well controlled until high levels but had less definition than was apparent in the rest of the audio band. This amplifier's greatest weakness though was the rather compressed character of the sound which seemed to make the music less dynamic than when heard through other well regarded amplifiers.

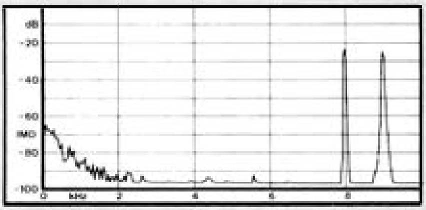
THE VERDICT

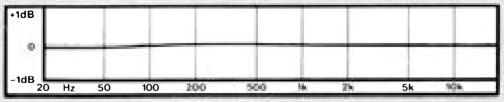
Quite an interesting amplifier and one which in many ways offers a lower cost alternative to Hitachi's successful HCA7500/HM7500 pre/power combination. The sound quality was quite acceptable, albeit rather lifeless, and the overall design showed signs of being well thought-out.

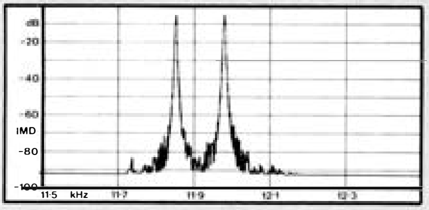
TEST RESULTS

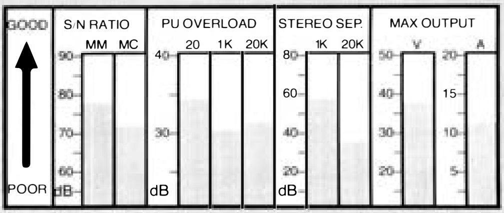
TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	99	102	99
	8+8Ω	87	92	89
	4Ω	116	126	126
HARMONIC DISTORTION (% @ 60 WATTS)		0·005	0·005	0·017

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	5Hz	77kHz





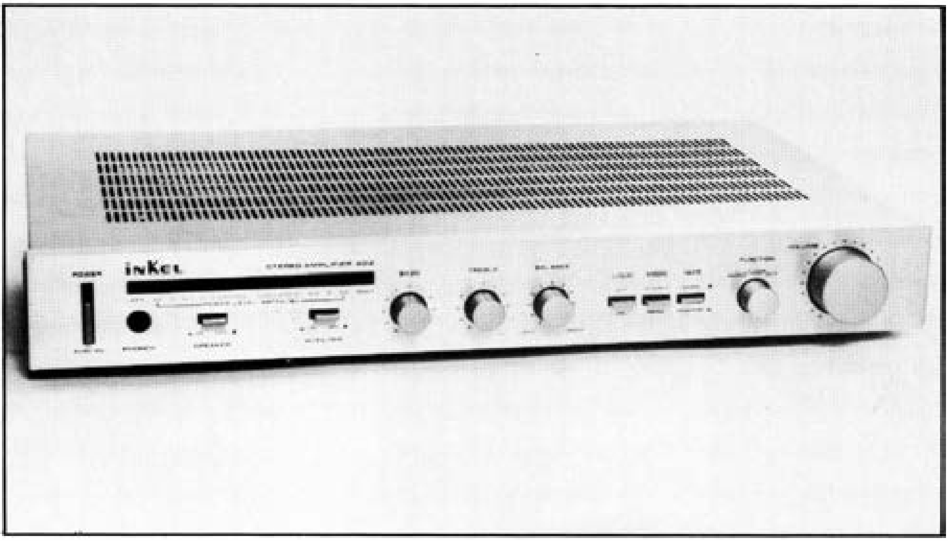




INKEL AD2

The Inkel brand of hi-fi equipment is the first popular range to come from the rapidly developing nation of South Korea — a country that seems well placed to replace Taiwan as a source of low-cost consumer electronics. The AD2 is a 25W integrated amplifier of slim dimensions, neatly styled with an anodised front panel and a grey steel cover. The knobs and switches are nicely finished with smooth actions, all the rotary controls being detented. Three inputs are provided, one of which is for moving-magnet pickups, as well as a single tape connection. Subsidiary controls are the Stereo mode switch, Loudness, Bass and Treble, Balance, and a rather simple HF filter with a 6dB/octave slope. Only one pair of loudspeakers can be connected but these can be switched-off by a front panel switch. There is also a headphones socket. The Inkel has attempted to provide a power-meter style display but the result is rather crude. For each channel there is a line of 6 LEDs which light up at levels equivalent to 3mW, 30mW, 100mW, 3W, and 50W (ref 8ohms). They must therefore, be considered as pretty lights fulfilling no really useful function.

One thoughtful touch, though, is the provision of speaker fuses on the back panel instead of hiding them away in the inaccessible insides of the amplifier.



CONSTRUCTION & TECHNOLOGY

The internal construction of the AD2 is well thought out and surprisingly neat. The wiring is very tidy but the PCB assembly has some components badly installed and a lot of residual flux left after the flow solder process. So obviously the boards are not cleaned after soldering. Power is provided by a substantial semi-C-core transformer which feeds a common power supply whose reservoir capacitors appear to be too small for good power delivery at low frequencies.

All the circuitry is discrete transistors arranged in very familiar circuits. The pickup-stage is a simple three transistor arrangement with the equalisation network wired in the negative feedback loop. The notable weak area in the preamplifier is the tone-control stage which uses a single transistor amplifier with Baxandall type tone-controls in the feedback loop. With the response set flat this stage works adequately, but on full boost the available feedback is reduced by over 10dB and the distortion rises significantly.

The power amplifier is of the familiar differential layout and is AC-coupled despite the Far East vogue for DC amplifiers. The output stage is fully complementary and I was very pleased to find that Inkel are using some very good Sanken power transistors which are extremely tough and reliable but which do cost more than other makes — definitely a surprise in a low-cost product. As a result the AD2 needs no protection circuits. the speaker fuses being quite adequate.

£84 **MERIT** ★★★☆☆ **VALUE** ★★★★☆

TEST RESULTS

No criticisms can be levelled at the Inkel for power output which well exceeded the manufacturer's rating. As expected the available power was lower at 20Hz because of the limitations of the power-supply but this should not be of great consequence in practice. The single channel power (8ohms) doubled the rated power and some 72W was available into 4ohms at 1kHz. The AD2 was also able to swing almost full volts into the capacitive load without undue distortion occurring.

The harmonic distortion was low for this class of amplifier and was found to be mainly third harmonic in nature.

The generous output capability of the Inkel was further emphasised by the recording of a 20amp peak current capability which demonstrated no operation of the protection circuits on high level transient signals.

Turning to the input characteristics the pickup stage was quite low-noise with good stereo separation particularly at 20kHz where it gave a far better result than many more expensive models. The equalised response was within an amazing ±0·2dB from 30Hz to 15kHz and I was so surprised that I had to recheck the result! At the frequency extremes the response is rolled gently down (20Hz −1·4dB, 20kHz −2dB). The pickup input capability was adequate although it did fall at high frequencies, a figure of 28dB being recorded at 20kHz.

It must be said that the Inkel performed very well in the laboratory giving a set of results far better than would have been expected from an amplifier of this price.

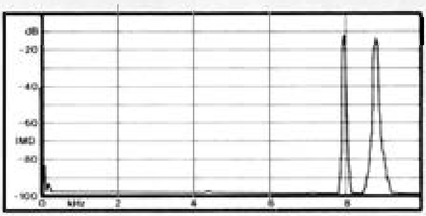
The AD2 also gave a good account of itself in the listening tests. Overall the sound was remarkably well integrated without any undue emphasis of any frequency band. A fair amount of detail was heard although it was thought that the stereo imaging was a little imprecise. Although the Inkel could reproduce music at fairly high levels it was an uncontrolled sound with a bass that lacked tightness and forcefulness. The high frequencies also became rapidly degraded, the initial slight grittiness becoming more pronounced as the volume was increased.

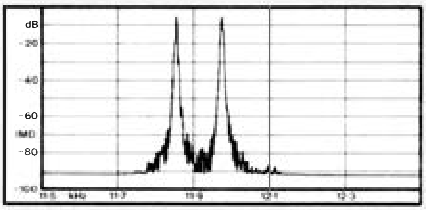
THE VERDICT

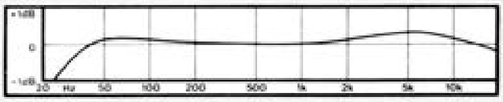
As much as I try not to, I do sometimes start a review with a few prejudgements. And this was the case with the Inkel for, let's face it, there have been some pretty ropey products imported from Korea in the past few years. But surprise, surprise, the Inkel isn't a bad amplifier at all. Its measured performance is almost suspiciously good for the price and it proved to be a reasonably nice amplifier to listen to as well. Maybe it wasn't too informative but neither was it unpleasant. Assuming large scale production shows some consistency of performance the Inkel could be a good buy.

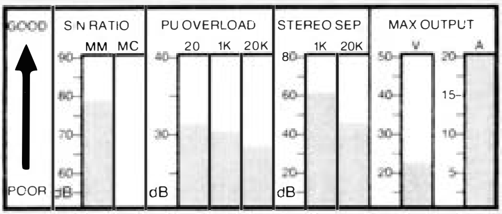
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	46	52	52
(WATTS)	8+8Ω	35	42	44
	4Ω	64	72	72
HARMONIC DISTORTION (% @ 25 WATTS)		0·030	0·015	0·033
FREQUENCY RESPONSE		LOW		HIGH
BETWEEN −1dB POINTS		6Hz		34kHz







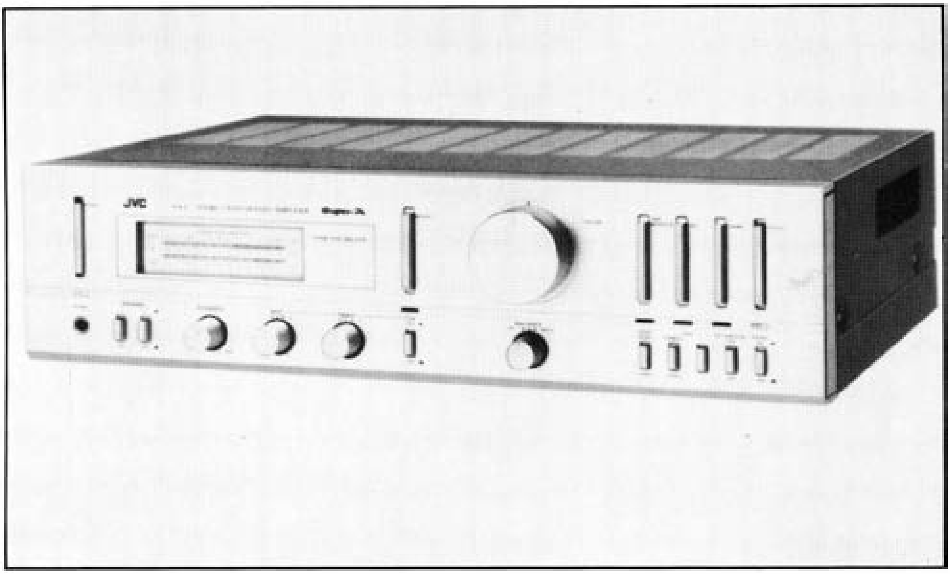


JVC A-X3

The A-X3 has a family resemblance to the more upmarket JVC models but its fascia is rather more cluttered and is visually dominated by an LED power-meter of the kind that almost all the manufacturers feel is necessary to ensure sales in this price range. The satin chrome finish is very smart, and to an extent the control layout is logically arranged. Four large push switches select the three main inputs and tape monitoring, while the smaller switches in the lower rank control stereo mode, tape dubbing (but only Tape 1 to Tape 2), Tape 1 or Tape 2 selection, subsonic filter, and a MC/MM selection switch. Also on the lower rank are the three tone-controls; bass, treble and one labelled loudness. This control alters the frequency response (mainly boosting the bass and treble) in the same way as a switched loudness filter but in this case the degree of compensation can be adjusted to 'taste'. The tone-controls can be switched out of circuit by another small switch and above this is the muting (−20dB) switch.

On particularly neat feature is the provision of several LED indicators (for confirming input selection etc) which are set into the groove which divides the upper and lower sections of the fascia.

The A-X3 is a moderately powerful amplifier (55W per channel) and has switching for two pairs of loudspeakers.



CONSTRUCTION & TECHNOLOGY

The JVC A-X3 has several PCB assemblies mounted in its steel chassis, each joined by complicated bunches of cable. The overall appearance is fussy although the component quality would suggest good reliability. A substantial extruded heat sink is fitted and particular attention has been paid to the safety of the mains wiring with much use of shrouding and extra insulation.

Also impressive was the use of screened cable for the signal lines, resulting in a lower level of hum and interference pickup and (all other things being equal) improved stereo separation.

Most of the preamplifier circuitry uses low-noise integrated circuits although the disc amplifier has dual low-noise FETs at the input. The power amplifier is a directly-coupled complementary design and uses one of JVC's special 'Super A' modules which puts all the complex dual-bias circuitry into a custom-made IC. This circuit ensures that the ouput stage is never biased off and therefore aims to minimise the effects of crossover distortion particularly at high frequencies.

£157 **MERIT** ★★☆☆☆ **VALUE** ★★☆☆☆

TEST RESULTS

The A-X3 had a generous output capability into 8ohms but when loaded by 4ohms the single channel *actually fell* from 87W to 56W (remember the output power from an ideal amplifier should double). The protection circuits operated to limit the output current so severely that, whilst a short 5amp peak could be sustained, more than 2½amps drawn continuously would trip the protection system. In consequence the A-X3 can only be recommended for use with the easiest of loudspeaker loads.

The harmonic distortion was very low at all frequencies but the HF intermodulation test did reveal some difference sidebands so obviously there is a degree of non-linearity at high frequencies.

By contrast the measurements made on the preamplifier stages were very good. Fine results were returned for the noise, pickup-input overload, stereo separation, and RIAA equalisation — the latter showing a slight HF boost.

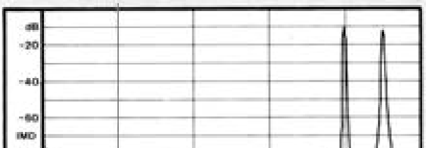
The listening tests revealed the A-X3 to be something of a disappointment. Although on rock and popular music the sound was initially very alive and exciting, extended listening showed the balance to be rather bright and the sound to be edgy and ultimately tiring. The bass, while powerful, was thought to lack definition with the result that the harmonics resulting from bowing a double-bass seemed to merge into an overfull 'bass-sound'.

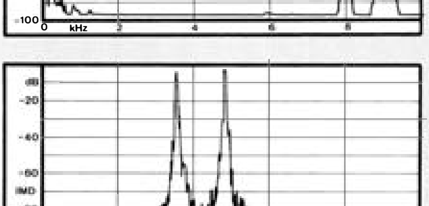
THE VERDICT

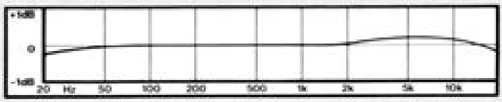
Good looking, good price, good facilities, if only the sound quality was as good as well. But then I don't suppose it will affect the sales, for the JVC A-X3 has a very smart appearance and the sound quality will, on the basis of a dealer's five minute demo, greatly impress the buyer with its power and attack. Used with a pair of small highish impedance speakers with a receding HF response, you'll probably achieve acceptable results but really there are better sounding if not better looking amplifiers available.

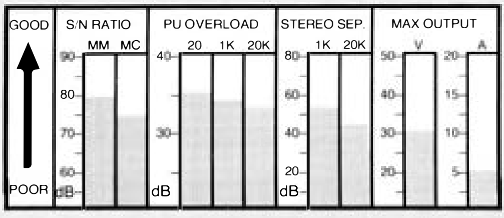
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	83	87	87
(WATTS)	8+8Ω	69	75	75
	4Ω	55	56	56
HARMONIC DISTORTION (% @ 60 WATTS)		0·004	0·003	0·005
FREQUENCY RESPONSE		LOW		HIGH
BETWEEN −1dB POINTS		3Hz		70kHz









JVC A-X5

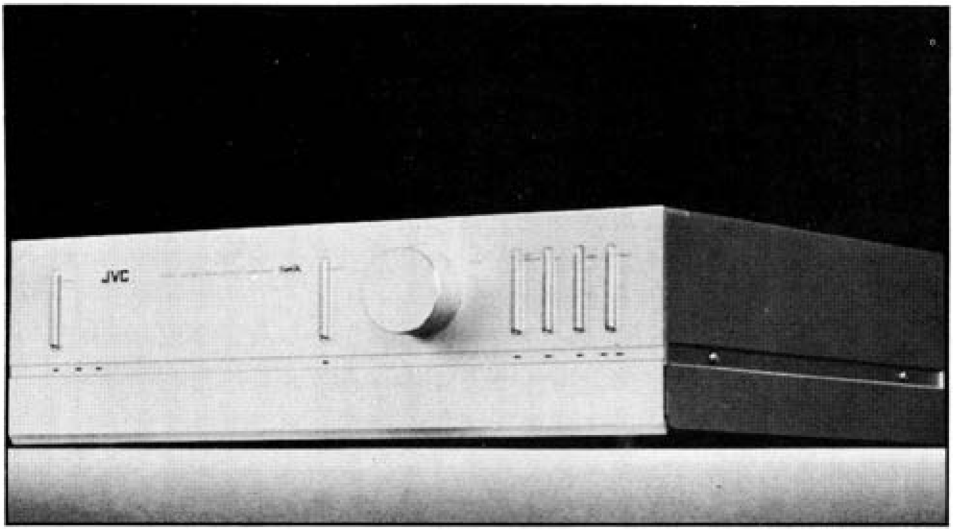
The A-X5 is a lower powered version of the A-X9, being rated at 70W per channel. It has the same elegant satin chrome finish front panel with a large smoothly rotating volume knob and a hinged lower section which can be opened to reveal all the subsidiary controls. The A-X5 incorporates JVC's 'Super A' non-switching power amplifier technology which attempts to bring some of the advantages of pure Class A operation without the attended disadvantage of low-efficiency and hence high heat dissipation.

This amplifier has inputs for both moving magnet and moving-coil pickup cartridges, as well as for two tape decks which can be switched to dub in either direction. One set of the tape connections is duplicated on the front panel, an arrangement which is extremely convenient if you just want to quickly hook on a second deck without the hassle of sorting out all the back-panel wiring.

The A-X5 has loudness and subsonic filters (about 18Hz at a rather too leisurely 6dB/octave slope), and bass and treble tone controls which can be bypassed by the tone switch to give a flat frequency response.

As with the other AX models, there is independent switching for two pairs of loudspeakers and a headphones socket.

Although it is quite elegantly styled, the A-X5 is a large amplifier and, with a front-to-back dimension of nearly 17ins, may well not fit many hi-fi shelves.



CONSTRUCTION & TECHNOLOGY

Removal of the cover of the A-X5 reveals a mass of circuitry broken down onto several PCBs interlinked by a large number of wires and cables. This amplifier seems to be substantially built with a large common transformer feeding a separate power supply (rectifier and reservoir capacitors) for each channel — this arrangement giving some of the benefits (eg reduced crosstalk) of twin power-supply designs but at a more economic cost.

The boards themselves are neatly loaded and are screened to label the component designations, and do include a number of test points that are no doubt essential when servicing such a complex design. The circuitry is mostly discrete in nature (indeed a very large number of transistors are used) and generally follows the same arrangement as the more expensive A-X9. It includes the 'Super A' twin bias circuit in the power amplifier stage.

£235 **MERIT** ★★☆☆☆
VALUE ★★☆☆☆

TEST RESULTS

Although the power output into 8ohms was well above the rated minimum, there was only a disappointingly small rise in output into 4ohms due to the operation of the current limiting protection circuits which also prevented the A-X5 operating with a 2ohm load. This low current capability was further demonstrated during the peak current test, a maximum of 5·5 amps being available but only for short periods — after half a second or so this figure fell to 3 amps when the protection limiter came fully into operation.

Otherwise the A-X5 recorded some remarkable results. Harmonic and IM distortions were virtually absent, and the disc stages were extremely quiet yet had excellent input overload margins and good stereo separation. The RIAA accuracy was within an amazing 0·1dB band from 20Hz to 20kHz — one of the most accurate responses I've measured in a commercial mass-produced amplifier.

Thus the measurements would suggest fine results provided that the loudspeaker demands little current from this amplifier.

However, in the listening tests the A-X5 was thought to produce a rather hard clinical sound having much in common with its metallic appearance. Despite the flatness of its measured response, there was a definite 'brightness' evident in the sound quality. Complex source material quickly became muddled and confused at any but low listening levels. Indeed it must be said that the A-X5 was not very pleasant to listen too for extended periods.

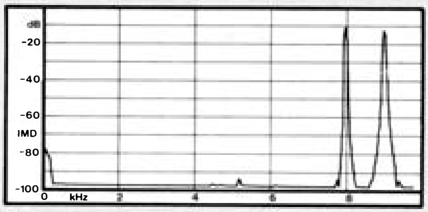
THE VERDICT

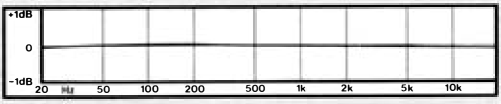
As with the other models in the AX family, this one has got the looks and the facilities. But somewhere along the line they forgot about getting the sound quality right. The A-X5 was tiring to listen through when using our choice of ancillary equipment and would be better partnered with a 'dull' pair of speakers and a rather less detailed source such as its companion JVC turntables. It's all a matter of what you most want from an amplifier.

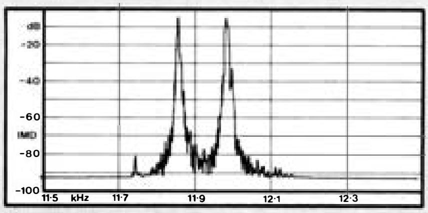
TEST RESULTS

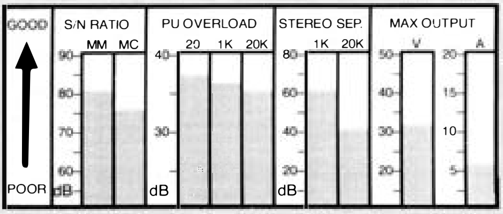
TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	94	96	95
(WATTS)	8+8Ω	79	84	84
	4Ω	92	100	100
HARMONIC DISTORTION (% @ 70 WATTS)		0·003	0·002	0·007

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	7Hz	76kHz









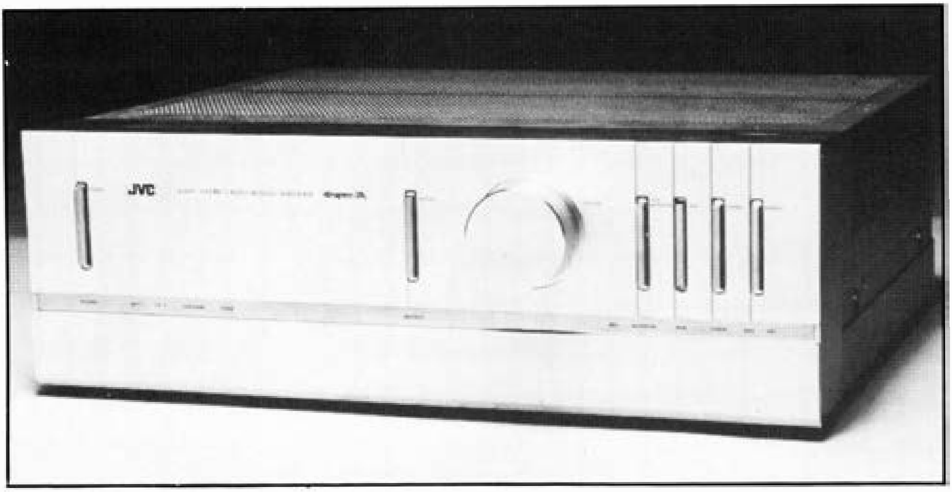
JVC A-X9

The AX-9 is the largest of JVC's AX range and carries a manufacturer's power rating of 105W. It is a big amplifier but beautifully finished in satin-silver and an appearance which, whilst starkly simple, is both imaginative and pleasing. User-convenience is high with a straightforward input selection by one of the four long push switches and level control by the large graduated volume knob. The remaining two push switches control muting and power.

However, when the lower front panel is hinged open a whole array of facilities is revealed. Switches select matching for moving-coil or moving-magnet cartridges (the latter having a choice of load impedances), tape monitoring and dubbing (in both directions), subsonic filter, loudspeaker switching (2 pairs) and tone defeat. The bass, treble and balance controls use tiny knobs beautifully engraved with graduated scales but rather too tiny for my hands. Also squeezed into this confusing panel are two pairs of phono sockets to permit easy connection of the occasional tape deck without the need to go ferreting around the back.

Useful though these extra facilities are I feel many owners will be content to set the subsidiary controls up once and for all, and close the flap after which the AX9 becomes a joy to use.

In common with the other amplifiers in the range this one uses the JVC 'Super Class A' technology which in reality is just a dual biasing network to prevent the output stage ever being driven into cut-off.



CONSTRUCTION & TECHNOLOGY

The chassis of the AX9 is extremely crowded and built in two layers. The top layer carries a massive toroidal power-transformer and two large extruded heatsinks carrying the output devices. Underneath are the power supplies and the main circuit PCBs. These boards are neatly loaded with good quality components but overall the chassis has an untidy appearance because of the large amount of wiring which is threaded to and fro around all the major components

In principle the circuit design of the AX9 is quite simple there being a disc amplifier followed by the selector switches and volume control feeding the power amplifier. That disc amplifier is in fact very complex using no less than 28 transistors and FETs and one IC in each channel. The circuit is essentially a differential arrangement having switched gain and input loading to suit both moving-magnet and moving-coil cartridges. A DC servo (feedback) amplifier is used to ensure absolute stability of the DC coupled circuit. The power amplifier is equally complex and incorporates JVC's 'Super A' circuit which is a dual bias system which ensures that the complementary output stage is never in a cut-off condition. This minimises the crossover and high frequency switching distortion.

£599 MERIT ★★★☆☆ VALUE ★★☆☆☆

TEST RESULTS

Although the AX9 could deliver a high power output into 8ohm loads, the output into lower impedances was curtailed by current limiting, the figures being 124W into 4ohms and only 61W into 2ohms. This limitation was further identified in the peak current tests, 3 amps being sufficient to trip the rather sensitive protection relay, the maximum of 20 amps only being possible for a fraction of a second. These measurements would suggest that it would be unwise to simultaneously connect two pairs of speakers to the AX9.

Harmonic and IM distortions were both very low and a spectral plot showed the JVC's circuit is quite effective in reducing crossover distortion.

Other measured parameters gave good results; the noise levels being especially low, even with the moving-coil input selected. One outstanding result was for the stereo separation which was a very high 65dB at 20kHz.

The RIAA equalisation accuracy was also superb amongst the very best that I have measured. From 20Hz to 5kHz the response was within less than ±0·1dB; the error for the most part being of the order of 0·05dB — close to the resolution of the best test equipment.

The listening tests, though, resulted in rather less praise. The sound quality was rather bright and metallic and despite an apparent clarity it was obvious that much low-level detail was being masked. The sound could be delivered powerfully but was never thought to be pure. Really an amplifier more suited to a party than a relaxed evening with your favourite recordings.

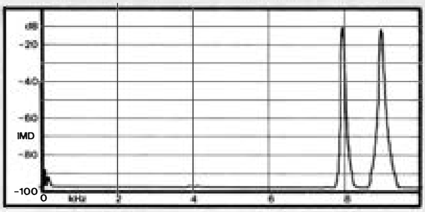
THE VERDICT

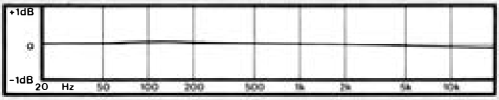
What a pity. The AX9 is full of original concepts both in its styling and its circuit design. Furthermore the standard of finish and overall appearance is excellent. The measured performance of many parts of the circuitry is also good, although there are inadequacies in the design capabilities of the output stage. These doubts were confirmed by the poor results of the listening tests which should be borne in mind by any potential purchaser.

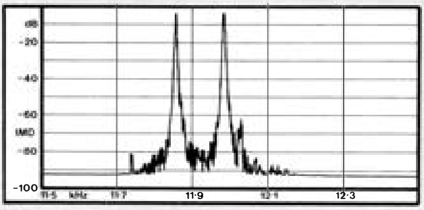
TEST RESULTS

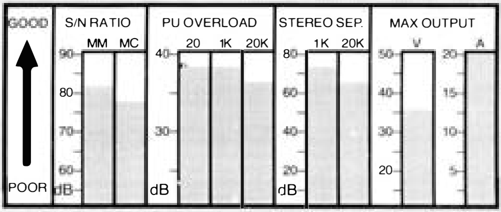
TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	137	137	132
	8+8Ω	123	122	119
	4Ω	106	110	106
HARMONIC DISTORTION (% @ 100 WATTS)		0·007	0·003	0·006

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	3Hz	41kHz









LENTEK

The Lentek amplifier is an unusual product in many ways. First, its 'Art-Deco' appearance which, despite some initial misgivings, does appear to be popular. Secondly, it is one of the few integrated amplifiers with audiophile pretensions. The price, too, raises a few eyebrows, for £750 for a nominally 60 watt integrated model is somewhat above the going rate.

The Lentek is a 'straight line' model with no tone controls, although there is a useful 12dB/octave low-frequency filter which turns over at around 20Hz and works on the disc inputs only. Four inputs are provided (including MM and MC disc sources) and two tape decks can be connected, the tape switching permitting cross-dubbing in both directions.

Three large push buttons switch the power, the low filter and the mono mode, and each has an adjacent LED to confirm the function. These switches look smart and obviously are expensive to produce, but their operation is rather 'clunky', especially as their appearance suggests a light action.

All the input/output connections are mounted horizontally so that you can see what you're doing without the need to turn the amplifier around. All the low-level connectors are gold-plated phono jacks whilst the single pair of loudspeaker outputs uses large binding posts ideal for heavy gauge cables. Adjacent to each of the disc inputs is a 5-gang slider switch which can give alternative cartridge loadings by changing the input resistance and capacitance.

The pre and power stages are split and so can be used separately although normally they are connected by two links.

One novel feature (previously seen on Amcron power amps) is an error-indicating LED on the front panel. The input and output signals are continuously compared and any difference equivalent to more than 1% causes the LED to flash.

CONSTRUCTION & TECHNOLOGY

This amplifier is built like a tank and much effort has gone into ensuring consistent and reliable performance. As a result there has been a degree of engineering overkill but many will believe this is money well spent

The Lentek is built as separate pre and power amplifiers on the same base-plate, with the two power modules behind the front panel. Between them is a large toroidal transformer rather untidily wrapped in a mild steel screen. The rear pre amplifier section is linked to the front panel through an ingenious array of extension rods and flexible couplings.

Generally, the workmanship is good, although some of the wiring is untidy in places.

The circuits are discrete bipolar transistor using an op-amp configuration throughout. Work has been done to make the open-loop performance of each stage fixed and predictable so that it does not vary with the characteristics of the particular batch of transistors used. The use of electrolytic capacitors in the signal path has been avoided so that at inter-stage coupling is through 2.2μF polyester capacitors.

The power amplifier has a protection limiter but the circuit has been arranged to give a bistable operation with a precisely defined operating point.

£785 **MERIT** ★★★★☆ **VALUE** ★★☆☆☆

TEST RESULTS

The midband power output was comfortably over the rated figure and hardly changed between single and both channel operation — a testimony to the size and performance of the power supply. The output power almost doubled into 4 ohms and continued rising into lower impedance loads although the protection limiter prevented the amplifier from fully driving the 2ohm load.

The power bandwidth was limited by the onset of distortion at 20kHz. So, for example, the maximum undistorted output at 1kHz into the 8ohm load was 87 watts but at 20kHz this fell to 78 watts.

A very good figure (greater than 200 at all audio frequencies) was recorded for the damping factor. The other measurements were all quite satisfactory, the low headroom at 20kHz being explained by the use of passive HF equalisation in the disc amplifiers.

The listening tests showed the Lentek to have a particularly neutral, uncoloured quality with the midband being particularly liked. The initially flat character was deceptive, for extending listening suggested that this amplifier was more accurate than several of its apparently more dynamic competitors.

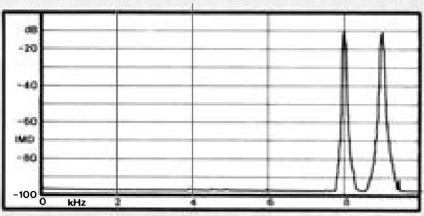
Strangely enough, the Lentek did not seem to be as powerful and effortless as the measurements suggested, and on some music the amplifier became muddled and plainly got into difficulty even though the IOC LED was not flashing.

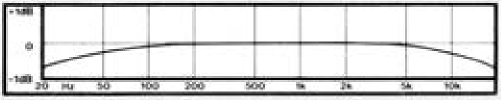
THE VERDICT

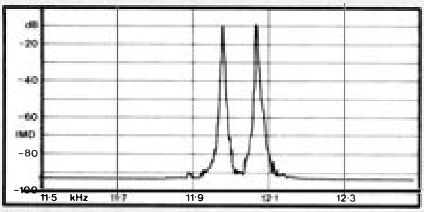
Built like the proverbial battleship and capable of some very good results. The sound quality was well-liked at low and moderate levels but the Lentek didn't have the effortless power its design suggests. It is expensive, but well built and potentially very reliable, but doesn't offer particularly good value for money. That said, the Lentek does offer an interesting alternative to some of the other far more overpriced audiophile amplifiers.

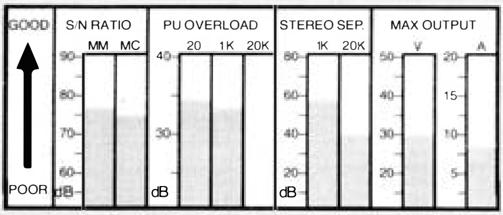
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	82	87	79
(WATTS)	8+8Ω	80	86	76
	4Ω	138	144	127
HARMONIC DISTORTION (% @ 60 WATTS)		0·012	0·008	0·034
FREQUENCY RESPONSE		LOW		HIGH
BETWEEN −1dB POINTS		8Hz		30kHz









LUXMAN L114A

The L114A is a budget priced amplifier from Luxman, a company perhaps better known for its more esoteric products. In appearance this amplifier is smartly finished with satin fascia but some of the minor controls might be thought to be untidily laid out. This Luxman has three main inputs and two for tape with switching for dubbing in both directions. The bass and treble tone controls are separate for each channel, instead of the more usual ganged controls, but although there is no bypass switch the centre (detented) positions were found to give a flat response. There is a low frequency filter but this is of limited usefulness for it comes in at 70Hz and is only of 6dB/octave slope. There is also a high filter of similarly slow slope, which comes in at 7kHz, and the usual loudness control.

Two pairs of loudspeakers can be connected and switched by two small push buttons; there is also a headphone socket.

All in all a smart little amplifier which looks far better in the metal than in a photograph.

CONSTRUCTION & TECHNOLOGY

Most of the circuitry is on a single large PCB which even has blank space for Lux to screen on a system diagram of the amplifier presumably as an aid to the service engineer. A single transformer powers the common power-supply which is extremely well protected with no fewer than five fuses, plus one for the mains primary.

The construction is very neat obviously helped by the fact that there are very few components used. No circuits were provided but an examination of the boards showed each disc amplifier to use just two transistors. There are no other active preamplifier stages, all signals being fed to a high-gain power amplifier which has the tone controls wired into its negative feedback loop. In many respects the L114A is an object lesson in value engineering!

£172 MERIT ★★★☆☆ VALUE ★★★☆☆

TEST RESULTS

This Luxman model proved to have generous reserves of power, even with both channels driving 8ohms some 65W was delivered at midband frequencies — some 50% up on the manufacturer's rating. Power into 4ohms rose to 100W, the recorded figure being limited only by regulation of the power-supply.

Harmonic and IM distortions proved to be adequately low showing little change across the audio frequency band.

Looking at the disc input an excellent figure was recorded for stereo separation, being 55dB at 20kHz. The input overload margin was also fine the drop at 20kHz probably being due to the use of the passive HF equalisation found on the more expensive Lux models.

One disappointment was the noise level which was rather high by current standards, although this noise was not troublesome in the listening tests.

The overall frequency response was very much curtailed outside the audio band, but some trouble was experienced in getting a flat (equalised) response on disc and even after 'tweaking' the tone controls there was still a degree of bass and treble roll-off.

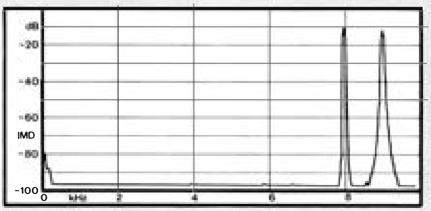
This amplifier was difficult to assess in the listening tests. Generally the sound quality was clear, with precise imaging. The initially 'middy' quality was counteracted by applying some bass and treble boost but the response still seemed lumpy. The sound was well controlled particularly in the lower registers and this control was preserved at higher levels.

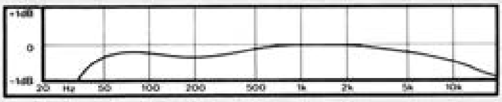
THE VERDICT

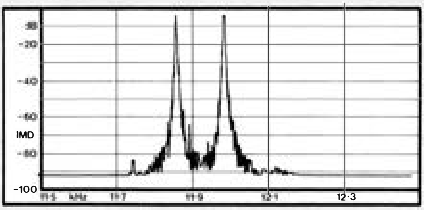
Quite a nice small amplifier which proved to be far more powerful than its 40W rating might suggest. Although let down by some tonal colouration it was also quite pleasant to listen to so to a large extent Lux have succeeded in producing a lower cost product having many of the attributes of their more expensive amplifiers.

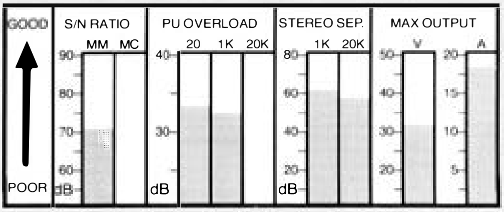
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	74	76	74
(WATTS)	8+8Ω	58	65	62
	4Ω	94	100	100
HARMONIC DISTORTION (% @ 40 WATTS)		0·02	0·019	0·03
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 19Hz		HIGH 22kHz









LUXMAN L116A

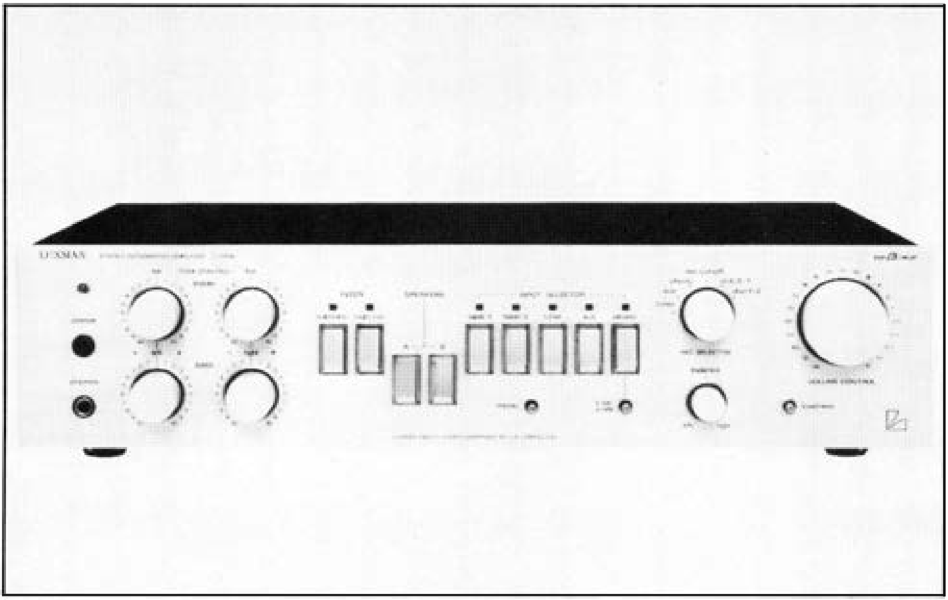
The L1 16A is a 70W integrated amplifier finished to the usual high Lux standards although on this occasion it must be said that the front panel layout is a bit of a mess with a variety of knobs, switches, buttons and indicators scattered about with gay abandon.

Inputs are provided for both moving-magnet and moving-coil cartridge sources and for two tape decks, with dubbing possible in both directions. Operation of the selector push buttons also causes an adjacent LED to illuminate to confirm the source.

As with the L1 14A separate bass and treble tone controls are provided for each channel. There are high and low filters whose effectiveness is limited because of their slow 6dB/octave slope, and there is a switchable loudness contour for the volume control.

Two pairs of loudspeakers may be connected and separately switched.

Both the disc and the power amplifier stages have Luxman's "Duoß" negative feedback, an arrangement where separate loops are provided for the DC and AC signal components so that the amount of feedback used can be optimised both for DC stability and for AC signal amplification.



CONSTRUCTION & TECHNOLOGY

Like the smaller L1 14A this amplifier is also built onto a single large PCB. But it does have a very large transformer feeding the common power-supply and a heatsink extending the full width of the amplifier and carrying the output transistors. The assembly is neat and to a good standard and screened cable has been used throughout for all signal paths.

The L116A uses a low-noise IC in the disc amplifier with switched gain and input loading when MC is selected. Again as with the L1 14A there are no other preamplifier stages, only a high gain power amplifier which has the tone-controls wired into its negative feedback loop. The power amplifier has a simple circuit with a complementary output stage, a bootstrapped voltage amplifier and a long-tail pair differential input stage. The amplifier has some DC-offset voltage protection provided by an output relay

£225 **MERIT** ★★☆☆☆
VALUE ★★☆☆☆

TEST RESULTS

Surprisingly despite its 70W rating the L-116A proved to have very little more available power output than its smaller (and less expensive) L-114A brother. Furthermore current limiting set in so that the output into 4ohms showed little rise above that delivered into 8ohms. The problem was further confirmed by the peak current measurement where only 7amps could be managed against the 18amps of the L-114A.

IM and harmonic distortion were very low, the latter showing a slight rise at high frequencies

Good results were returned from the measurements on the disc inputs although the midband stereo separation could have been better. Again some difficulty was experienced in obtaining a flat frequency response through the disc inputs (even after adjusting the tone-controls) the flattest curve still showing some bass and treble boost.

In the listening tests the L-116A proved to be a disappointment. The sound quality was quite pleasant at low levels with good revelation of detail and good imaging. But at higher levels the sound became uncontrolled the bass in particular losing its definition and precision.

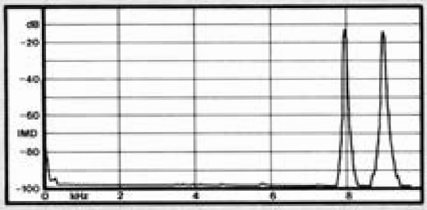
THE VERDICT

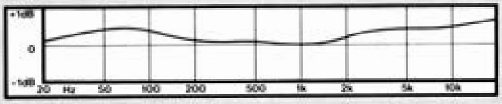
After evaluating the L-114A it is very difficult to see the more expensive L-116A as anything other than a disappointment. Subjectively the smaller amplifier proved to be just as powerful so the only benefit for the extra money is the moving-coil input. But against this is the comparatively poor sound quality which contrasted badly with the L-114A. Our recommendation is that you buy the cheaper model and spend the difference on an MC head-amplifier if you need it.

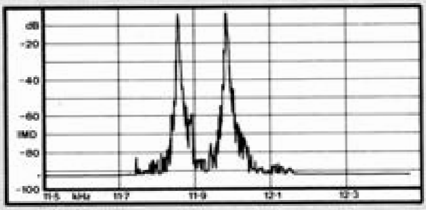
TEST RESULTS

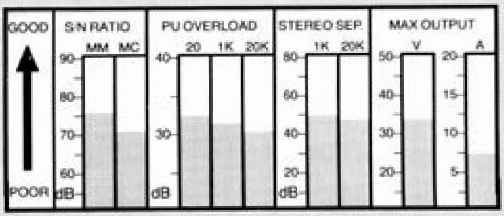
TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	92	97	92
	8+8Ω	78	83	78
	4Ω	96	100	100
HARMONIC DISTORTION (% @ 70 WATTS)		0·005	0·002	0·010

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	8Hz	68kHz









MARANTZ PM350

The PM350 is another remarkably inexpensive unit from Marantz, this model being rated at 38W per channel. It is immediately recognisable as a Marantz product; the gold coloured finish and chunky black lettering may not be to everyone's taste but is distinctive enough to be a trademark. The front panel layout is logical and, therefore, leads to straightforward use, but the mixing of rotary and slider controls is, to me, something of a styling disaster.

The three main inputs are selected by a rotary selector switch and only one true tape connection is provided although the auxiliary input doubles as a tape input for dubbing purposes. The PM350 has three tone controls, these being the sliders giving cut and boost of bass, treble and midrange frequencies. The tone controls cannot be bypassed but the zero positions are accurately detented. There is, of course, a loudness control and a subsonic filter — the latter not seeming too effective in use. Two pairs of loudspeakers can be driven and switched separately by push-buttons. Finally there is the undoubtedly eyecatching LED power indicating display. This display attempts to use effectively five indicators to display power output represented on an 8ohm scale of 5mW to 50W. It proved to be only roughly accurate and of little practical benefit, but if the competition has it then so must Marantz!

CONSTRUCTION & TECHNOLOGY

The PM350 is neatly built with most of the circuitry on one large board which extends from the front of the case to the rear. A large finned heatsink runs the length of the case and is quite generously sized for the power rating of this model.

Mains connection is through an approved 2-pin connector and moulded cable, the transformer being double insulated. The mains wiring is neat and safe although no shrouding has been used over the various connections so owners would be well advised not to remove the lid.

The circuitry of the PM-350 uses ICs throughout the preamplifier stages. A dual low-noise IC is used as a disc amplifier with another dual IC as a line amplifier wired to give a gain of about ×8. Conventional tone controls are then wired into the negative feedback loop of a third IC.

The circuit of the power amplifier is strangely old-fashioned although competently designed. A long-tailed pair input stage drives a common-emitter voltage amplifier which has a collector load bootstrapped to the output; this drives a Darlington type complementary output stage. Switch-on muting is performed by a transistor wired across the input of each power amplifier. The transistor is initially on (shorting out the signal line) and then turns off after the preamplifier stages have stabilised. The power amplifiers are AC coupled and the low-filter works by switching the value of the feedback decoupling capacitors. This technique gives only a slow 6dB/octave slope at an imprecise turnover frequency because the capacitors used have a ±20% tolerance on their value.

£89 **MERIT** ★★★☆☆
VALUE ★★★★☆

TEST RESULTS

The PM-350 proved to have a good output capability particularly on single channel operation. The power output rose substantially into a 4ohm load, a figure of 87W being recorded at 1kHz. This Marantz was also totally unaffected by the capacitive load. The fine current delivery was further confirmed by a figure of 22amps for the peak capability.

Harmonic distortion, though, was a little on the high side — certainly above the manufacturer's specification. The damping factor was also on the low side (but still adequate), a figure of about 30 being measured.

The preamplifier was also a moderate performer with noise, overload and stereo separation figures that were adequate but no better than adequate. The overall bandwidth was found to be sensibly limited at both ends, the −1dB response extending from 10Hz to 35kHz. The RIAA response on disc was reasonably flat above 1kHz but below that there was a definite upper-bass hump which seemed to be due to the tone controls not being exactly flat in their centre settings

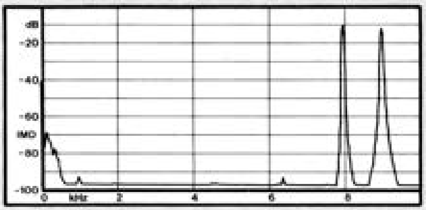
When the listening tests started, not too much was expected of an amplifier from this price bracket so we did not try to be hyper-critical. However, the PM-350 returned a creditable performance for, although the sound quality was somewhat coloured (possibly due to the lumpy response), it did seem to be well controlled. The character of the sound was thought to be smooth but lacking in any real detail and clarity, particularly at low and lower-mid frequencies. The Marantz was found to go quite loud without producing aggressive and brittle sounds, but the sound became more muddled than ever.

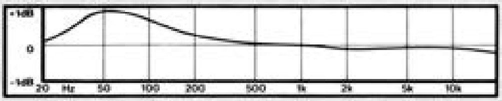
THE VERDICT

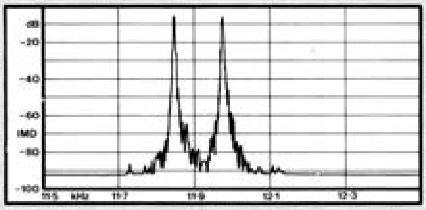
The PM-350 has only a moderate performance but did seen to be remarkably unaffected by loudspeaker loadings. The sound quality was such as to not reveal too much of the subtlety of the music yet it had a pleasantness that would be endearing to many. Really, if you've only got £89 to spend you're not going to go too far wrong with a PM-350 and you'll have ample opportunity to do a lot worse.

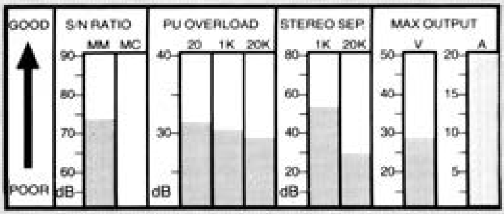
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	53	57	58
	8+8Ω	40	46	46
	4Ω	77	87	83
HARMONIC DISTORTION (% @ 35 WATTS)		0·39	0·2	0·1
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 10Hz		HIGH 35kHz







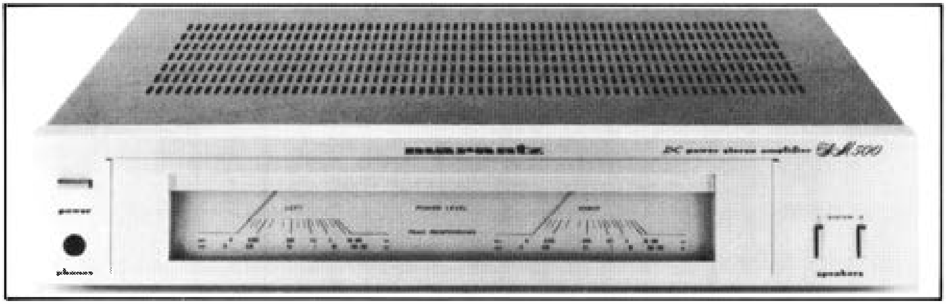


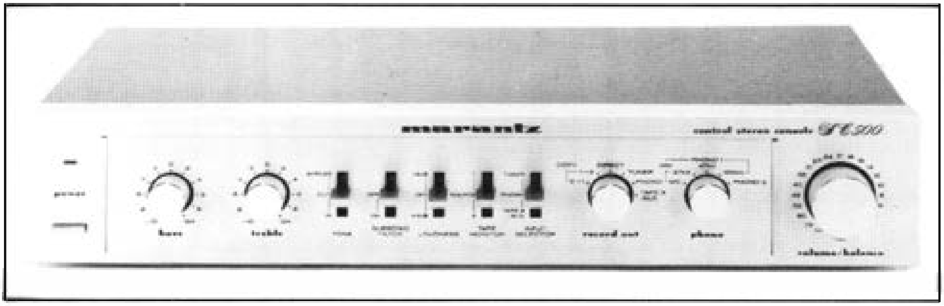
MARANTZ SC500 SM500

This combination is offered as an alternative to Marantz's range of integrated amplifiers and has an output rating of 60W per channel. The styling is almost predictable with its gold coloured fascia and case and black 'Marantz script' lettering. Although this formula works on many of their products I must say that I found the SC500 preamplifier to be quite ugly. This unit uses a concentric volume and balance control and has three main inputs, with switching for both moving-magnet and moving-coil cartridge sources; and also can be used with two tape decks, the switching permitting dubbing in both directions.

The bass and treble controls can be bypassed if required and there is a low-frequency filter, but it is one of those pretty useless 6dB/octave affairs. Finally there is a loudness switch, but it has two positions offering two degrees of boost to the frequency extremes to suit different speaker types.

The front panel of the SM500DC power amplifier is dominated by the two large illuminated 'power meters' which, I must admit, were far less offensive that the more modern flashing lights, indeed almost comforting to those of us with a background in recording. Apart from the power switch there is just a headphone socket and separate switches for the two pairs of loudspeakers. Like most Marantz products these two both use the 2 pin mains connector which is becoming increasingly common on consumer equipment but which I find doesn't always provide a firm contact.





CONSTRUCTION & TECHNOLOGY

Internally the construction of the SC500 is delightfully simple with one board soldered to the back-panel input/output sockets and another holding the front panel controls. The two are linked by ribbon cables and the whole chassis is neat if a little empty!

The circuitry is also simple with an IC disc amplifier (using two low-noise transistors at the input) and an IC line amplifier with passive tone-controls (most unusual these days). The remaining circuitry comprises the power supplies and a delayed switch-on relay for muting the output.

Like the preamplifier the SM500 is also very neatly laid out with a small PCB carrying the common power supply and the two power amplifier circuits. A very substantial transformer is used and a large heat-pipe dissipates the heat generated by the power transistors. Some care seems to have been taken to ensure a high standard of workmanship.

The circuit design of the discrete power amplifier is very simple, being the traditional long-tail input pair operating at constant current followed by a voltage-amplifier with dominant pole compensation and driving a constant current load. The output stage is fully complementary with V-I protection transistors.

Interestingly enough the SM-500 uses a form of DC servo in the feedback to ensure DC stability without compromising the AC signal performance.

TEST RESULTS

The power-output with both channels driving 8ohm loads was only just above the 60W rating but rose to a reasonable 110W or so driving the 4ohm load. For very short periods a peak current of 8 amps was measured but this could not be sustained for more than about one second after which a limiter clamped the maximum to 2 amps. This would mean that although this Marantz could handle transient signals into most loudspeaker loads, sustained signals such as organ pedal notes might cause problems.

The measurements of distortion suggested that it might be better to rate this model at 50W per channel at which power the distortion was very low.

As for the other measurements — well a generally good crop of results was recorded. Perhaps the disc input noise could have been lower and the frequency response have been less extended at the top end. There isn't any benefit to be had by having a response that stays virtually flat well above audio frequencies; beyond 120kHz in the case of this Marantz.

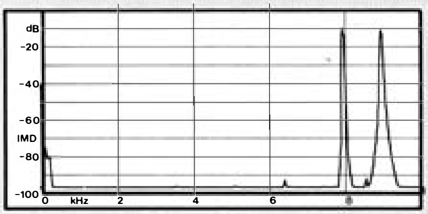
In the listening tests this Marantz combination gave a disappointing result especially when the sound quality of some of their cheaper integrated amplifiers is recalled. The sound was particularly felt to be muddled with a lack of definition. The bottom end was rather uncontrolled and not very precise; the harmonic structure on bowed bass strings merging into a low 'humming'. It wasn't the case that the sound was truly awful — just that there was little I could find to say in the way of praise.

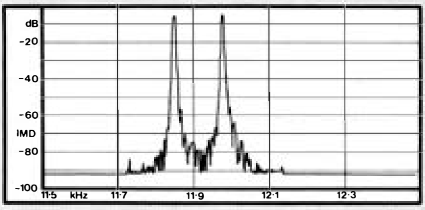
THE VERDICT

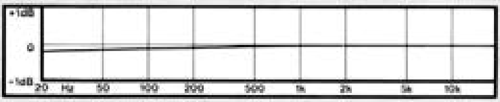
I have to ask what this Marantz pre and power combination does that their range of integrated amplifiers doesn't do better. Their PM-350 and PM-450 models offer near enough the same subjective loudness, better sound quality and you save £180 and up into the bargain. You could say that Marantz are their own worst enemy by making cheaper mean better on this occasion.

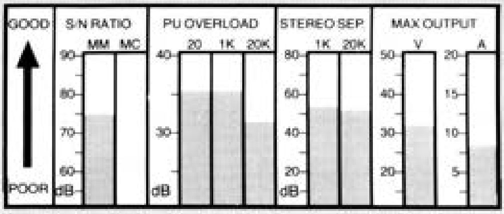
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	80	78	79
(WATTS)	8+8Ω	64	67	67
	4Ω	110	116	110
HARMONIC DISTORTION (% @ 50 WATTS)		0·007	0·006	0·009
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 7Hz		HIGH 120kHz

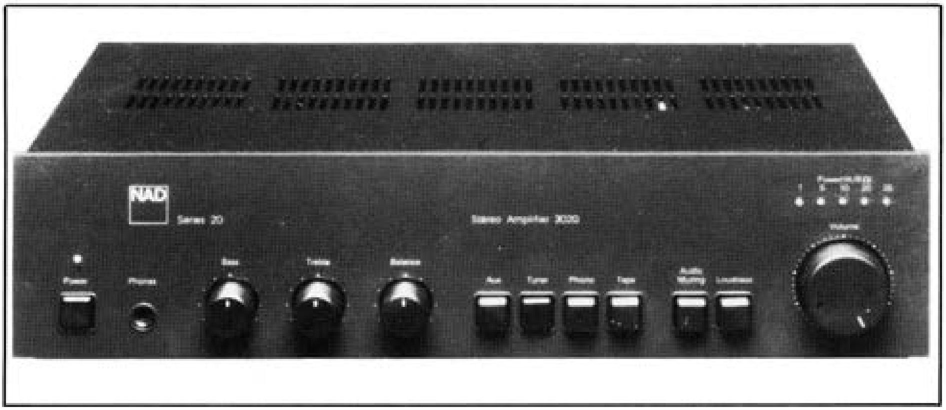








NAD 3020



The NAD3020 started life as a low-cost 20W integrated amplifier manufactured in Taiwan. But it wasn't long before it began to establish a reputation for having a quite passable sound quality compared to its competitors.

First impressions, though, aren't too good. The overall appearance is rather crude with the plastic front panel carrying ugly knobs and switches. The whole case seems flimsy and skimped compared to the average Japanese (or even Taiwanese) product.

Three inputs are provided, including one tape connection, and these are selected by a row of push buttons. A further two buttons give muting (a 20dB gain reduction) and loudness. The bass and treble controls cannot be bypassed and no switched filters are provided. NAD have provided a poor man's power meter which consists of a row of five LED indicators set above the volume control. At best they have little usefulness and at worst they are irritating. Terminals are only provided for one pair of loudspeakers but these are directly connected without the potential degradation of switches, etc. One potentially useful feature is the break between the pre and power amplifier sections through phono sockets connected by U-links. A second input connection to the power amplifier is marked 'Lab-IN' and bypasses the band-limiting filters for those occasions when it is used as part of an active system. There is also a switchable 'soft-clipping' circuit which is intended to moderate the sudden harshness that accompanies clipping; however its theoretical usefulness was limited by the reduction of apparent dynamic range when this circuit was in operation.

CONSTRUCTION & TECHNOLOGY

Well, to be frank, the NAD 3020 wouldn't win any awards for its constructional quality. The chassis and case are flimsy and inspire no confidence although, to be fair, not many of us drop our amplifiers that often. Almost all the circuitry is on a single large printed circuit board which is neatly laid out but not so neatly assembled; great splodges of flux being found on the top surface. A simple aluminium bracket does duty as the heatsink for the four metal-can 2N3055/2955 power transistors.

The amplifier has adequate fuse protection but the transformer is not double insulated so the earth-wire of the 3-core mains cable must be connected for safety.

The circuit design of the 3020 is innovative for a low-cost product. The disc stage uses a novel sort of long-tail pair inputs, but with the emitters left floating, and a push-pull output stage. The voltage gain stages of the power-amplifiers are powered from a stabilised supply rail so that the supply fluctuations of the output stage are isolated to a degree. Each output has a thermal contact breaker in line as an overcurrent protection but to minimise any signal degradation the negative feedback is partly taken from the other side of this switch so that the distortion is reduced in accordance with the laws of feedback.

The 3020 has in the past proved capable of delivering high peak currents into a load but again this particular sample only delivered 16amps (still an excellent figure) whereas other samples I have tested have been capable of 20amp peaks.

A single power supply feeds both channels and uses reservoir capacitors which are too small to ensure a good low-frequency power-bandwidth. However the saving on the cost of these parts is compensated for by the 2N3055 power transistors, the use of which is exceedingly generous in a 20W amplifier.

TEST RESULTS

The 3020 proved to have a good power-output capability, the single channel (8ohm) power being 44W although this fell at 20Hz to 39W. The power into 4ohms was limited by the tripping of the protection circuit at about 60W output; nonetheless this is a good figure and I should mention that I never experienced such protection operation on other samples of the 3020 that I have measured in the past.

Harmonic distortion was low right across the audio band as was the intermodulation distortion. In context of the price the performance of the preamplifier section was very good, the main weakness being the irregular RIAA equalisation, which overall showed a degree of bass-cut and treble-boost. Trimming the tone-controls helped but a 'flat' response could not be achieved. The auxiliary input showed sensible band-limiting, the response being down 1dB at 17Hz and 30kHz.

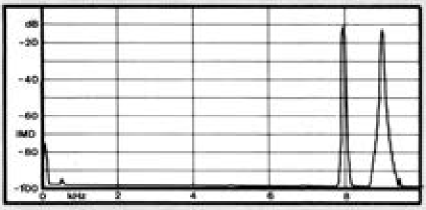
On audition the 3020 produced what could be described as an exciting sound with good rendition of dynamics. Some detail was felt to be missing in the midband but nonetheless the sound had good clarity. The 3020 did not reproduce bass instruments very well, particularly at higher volume levels, but overall the sound quality was a cut above that normally served up by a low-cost amplifier.

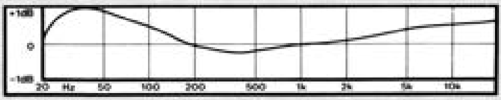
THE VERDICT

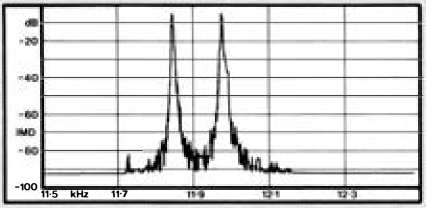
To read some magazine reviews it might be thought that the NAD 3020 is the last word in audiophile amplifiers which it patently is not. But for the price it gives an excellent account of itself and shows that some intelligent work has been done at the design stage. The 3020's sound quality will find favour with most listeners but to be fair it now no longer stands alone in its price bracket. But if your budget is tight this is still one for your short list.

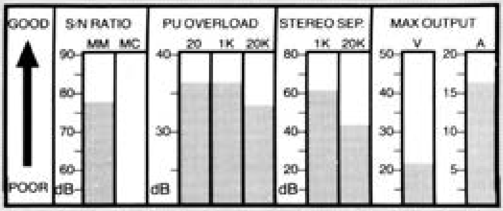
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	39	44	42
(WATTS)	8+8Ω	28	36	34
	4Ω	56	58	56
HARMONIC DISTORTION (% @ 25 WATTS)		0·018	0·004	0·016
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 17Hz		HIGH 30kHz

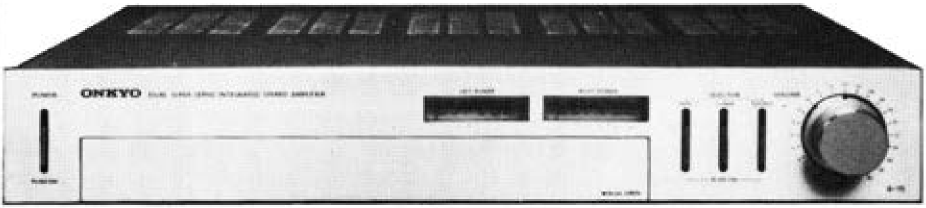








ONKYO A15



The Onkyo brand is little known in the UK but is now being imported from Japan by Goodmans Ltd. The A15 is a 30W per channel integrated amplifier of quite slimline dimensions and finished in silver grey with a matching anodised aluminium front panel. At first glance the Onkyo looks remarkably devoid of controls and facilities, the only visible parts being power and selector switches, volume control and a pair of largely ineffectual power meters (analogue types) which attempt to read from 10mW to over 100W (equivalent into 8ohms) — there's optimism for you! However the A15 has a neat front panel flap which opens smoothly and easily to reveal a number of subsidiary controls including bass, treble, and balance controls plus push buttons for loudness and tape monitoring (tape 2). Of the three normal inputs the auxiliary also doubles as tape 1 if it is desired to connect two tape decks. There is no tone-cancel switch nor are any of the tone controls (or balance) centre-detented so the frequency response flatness will depend upon the accuracy of those control settings. Although two pairs of loudspeaker terminals are provided, the front panel switch is a push button which gives A or B speakers on but not both. To do that you need to wire the speaker cables differently (as explained rather badly in the owner's manual) after which the switch can offer A or A + B combinations. There is also a headphone socket which disconnects the loudspeakers when the plug is inserted.

CONSTRUCTION & TECHNOLOGY

Inside, the circuitry is built around a strong, rigid steel chassis. A large transformer is used and this is shielded by a steel box formed by a number of chassis members. A substantial extruded heatsink is fitted and this has attached a thermal cutout to protect the amplifier in the event of overheating. There are internal mains and speaker fuses; the latter obviously inconvenient from the user's viewpoint especially when the case carries a printed warning not to pry inside! Overall, although the amplifier is well made, the construction is rather messy with bits and pieces seeming to be hung everywhere.

No circuit information was provided but it was found that all the signal amplification was by means of ICs and a large power IC contained both power amplifiers. The front panel legend makes much of Onkyo's 'Dual Super Servo Circuit' but there was no explanation in the manual. As far as I could deduce this phrase refers to a separate DC amplifier used in the negative feedback loop to maintain good DC stability without requiring the same order of feedback at audio signal frequencies.

£99 **MERIT** ★★☆☆☆ **VALUE** ★★☆☆☆

TEST RESULTS

For a nominally 30W amplifier the A15 was found to be generously rated with a single channel capability of 47W into 8ohms. The power rose nicely into a 4ohm load until power supply regulation curtailed the output at about 59W. No problems were experienced with the capacitive load either. Harmonic distortion was adequately low although it rose at the frequency extremes.

The peak current capability of the Onkyo was above that of some of its Japanese competitors but the 10amp limit was determined by the protection circuits.

The measured damping factor was very high which suggested that a high amount of negative feedback is used around the power amplifier.

The input measurements revealed sensible sensitivities for both tuner and pickup inputs although the measured noise was disappointing, especially on the tuner input which was hardly less noisy than the pickup input. Pickup overload and stereo separation figures fell into the just adequate class and an improved crosstalk figure at 20kHz would definitely be preferable.

The RIAA equalisation result was rather dependent upon the tone-controls whose zero settings seemed to be inaccurate. Some juggling eventually gave an overall ±0·6dB over the midband but with some premature roll-off of both the bass and the extreme high frequencies.

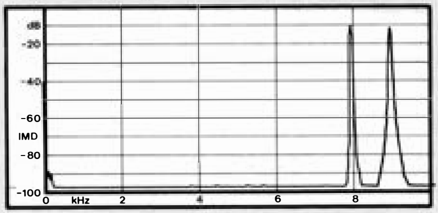
The Onkyo sounded a little 'middy' when first auditioned but a tweak of boost from both bass and treble controls give a subjectively flatter response. First impressions suggested that the sound quality was poor with a hardness spoiling the high frequency reproduction and a wallowing uncontrollable bass. However, when changed to an efficient pair of speakers the volume could be reduced to give a more pleasant sound quality. Even so the overall feeling was of a muddled sound which seemed to constrict the dynamics of the music.

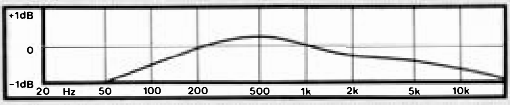
THE VERDICT

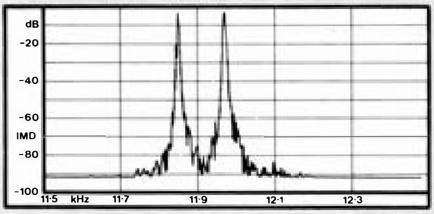
A very smart little amplifier finished to a high standard and offering a good package for the price. Technically one or two problems reared their head, particularly the flatness of the frequency response or, to be more correct, the lack of flatness! Similarly the sound quality was unimpressive although at no time was the sound particularly unpleasant. The Verdict? A middling amplifier in an upmarket box. Buy one and fool your neighbours.

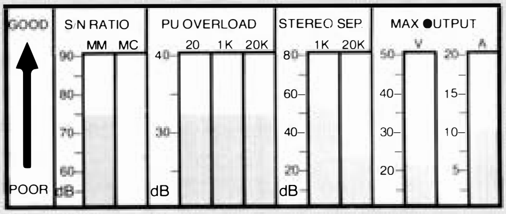
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	46	47	46
(WATTS)	8+8Ω	38	42	42
	4Ω	56	59	59
HARMONIC DISTORTION (% @ 30 WATTS)		0·025	0·012	0·032
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 70Hz		HIGH 33kHz







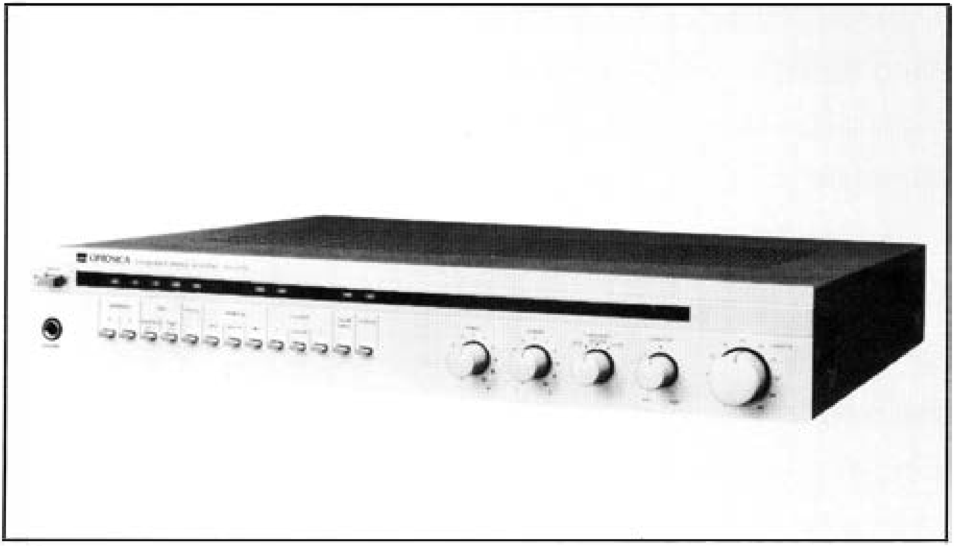


OPTONICA SM-5100H

The SM-5100H is Sharp's middle-of-the-road model nominally rated at 40W per channel and commonly seen in their rack systems. This amplifier is attractively styled to match their other products and has a distinct 'high technology' appearance. Almost half the front panel is taken up by a row of 13 identical miniature push buttons most of which operate a LED indicator behind the clear plastic strip running across the fascia. At first this array can be quite confusing, constant reference to the printed legends being necessary.

The Optonica has the usual three inputs but no moving coil option. The two tape deck connections can be switched to give dubbing in both directions. Switches are provided for muting (−20dB), loudness, subsonic (30Hz) and high (7kHz) filters, and stereo mode (a facility that is now found on fewer and fewer amplifiers). The bass and treble tone-controls cannot be bypassed but their centre-position proved to be fairly accurate.

The usual headphone socket is fitted as are two sets of loudspeaker connections, each of which is switched. One nice extra is the separation of the pre and power amplifier stages which are normally linked by a pair of removable U-links plugged into the phono sockets.



CONSTRUCTION & TECHNOLOGY

This Optonica is built to the usual competent Japanese standard although some of the wiring is a little untidy. The PCBs are neatly loaded and some of the larger components are glued to the boards to prevent any strain on the soldered joints.

A large toroidal transformer is used and its already low external magnetic field is further reduced by enclosing it totally in a steel cover.

The disc amplifier stage is on the rear panel PCB which also carries the input sockets, a remotely operated switch being used for selection. The rest of the circuitry is on a board positioned between the front panel and the full length generous sized heat sink.

The circuit arrangement is fairly conventional with a discrete transistor preamplifier and using large integrated circuits in the power amplifier. The disc amplifier is a simple three-transistor affair with a low-noise input FET and includes some useful RF filtering. The RIAA equalisation is achieved by a network in the negative feedback loop. The second stage uses a similar three-transistor circuit for the feedback tone-controls. The power amplifier couldn't really be much simpler with a voltage amplifier IC driving a power IC. These integrated circuits also include comprehensive protection circuits.

£125 **MERIT** ★★☆☆☆ **VALUE** ★★★☆☆

TEST RESULTS

The Optonica exceeded its maximum rated output by a small margin giving 46W with both channels driving 8ohms. The power into 4ohms increased to 72W and although the output fuse blew when driving the 2ohm load it was clear that there was plenty of current available and that it was not being prematurely limited by protection circuits.

The harmonic distortion was low in the midrange but rose at high frequencies, the 2-tone IM spectra showing quite a few sidebands of significance.

The measurements taken of the preamplifier stage gave good results for noise level, disc overload margin and stereo separation. The equalisation accuracy was adequate over most of the audio band but a slight lift occurred in the 80Hz region followed by a rapid bass roll-off.

The filters were found to be only token efforts the high filter only having measured 5dB/octave slope. The subsonic filter was also a one pole network (6dB/octave) and pulled the response down 1dB at 100Hz; 3dB at 50Hz and nearly 13dB at 10Hz — unfortunately having a quite significant effect in the audio band.

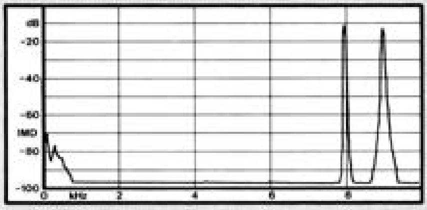
Unfortunately the Optonica did worse than the average in the listening tests. It was very much a case of the sound quality being fine as background noise but in hi-fi terms it just wasn't very satisfying to listen to. Particular weaknesses were the imaging which was vague and lacking any sensation of depth; and the bass which lacked both definition and solidity.

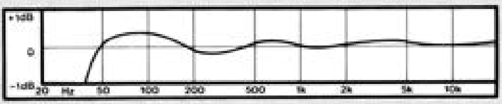
THE VERDICT

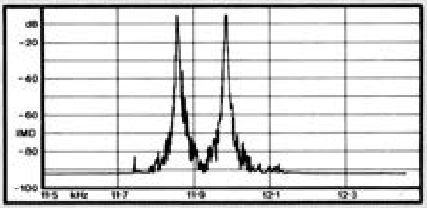
This Optonica is at first glance a quite attractive product offered at a very fair price. The laboratory tests revealed a balanced performance and many of the results were good for an amplifier in this price bracket. But the sound quality was thought to be well below average and unlikely to give much satisfaction to the serious listener. On the other hand if you just want an amplifier to reproduce Radio One while you do the washing up, well look no further.

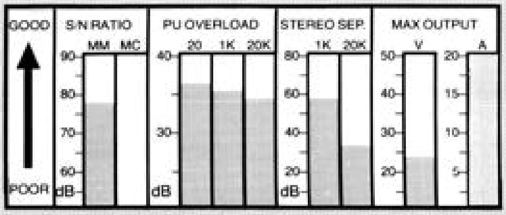
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	50	52	50
(WATTS)	8+8Ω	46	46	46
	4Ω	71	72	72
HARMONIC DISTORTION (% @ 40 WATTS)		0·09	0·06	0·1
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 13Hz		HIGH 30kHz









PS AUDIO VK M2

These units are typical representatives of American enthusiast-orientated product. PS Audio originally started as a kit manufacturer and the origins are still evident in the rather utilitarian styling and finish of the black boxes. This functional rather than pretty philosophy continues with the bare bones modular concept enshrined in the system which comprises three units: the phono preamplifier, the VK volume control, and the M2 power amplifier. Other than a supply LED the front panel of the preamp is blank, and at the back there are single pairs of phono type input and output sockets and a mains supply cable.

The ultimate is a straight-line system, perhaps, but someone realised that a volume control could have its advantages so along came the second box, the entirely passive VK volume control. There is, in fact, a separate control for each channel together with a source switch as a further concession to practicality to allow the alternative connection of a tuner and tape source.

The M2 power amplifier is again fairly basic offering 40W per channel although it can be wired in a bridged configuration to a give a claimed 160W mono into 8ohms. Once again the accent is on simplicity (or perhaps low-cost), the plain black boxes having just a power-switch and LED supply indicators on the front panel.

CONSTRUCTION & TECHNOLOGY

Well, to be honest the PS equipment wouldn't win any prizes for the quality of its construction. They have used good components (glass-fibre PCBs, metal-oxide resistors, and Switchcraft audio connectors) but the crude metalwork and basic 'home-built' appearance are initially off-putting (particularly the use of slotted screw-holes to take up the wide tolerances). However, a close inspection did reveal neat loading of components onto the boards and competent soldering and wiring.

The preamplifier transformer seemed to be rather small and did buzz slightly (although the electrical hum level was very low), but an optional high-current power unit is available from PS.

The circuit design has some interesting features, the preamplifier having two almost identical gain stages (designed with an Op-Amp configuration) separated by an entirely passive RIAA equalisation network which provides the necessary bass-boost and treble-cut. This use of passive equalisation means that the linear gain stages need far more headroom than is the usual case because there is no gain reduction by the application of negative feedback. For this reason the preamplifier operates from ±25 volt supplies.

All electrolytic capacitors are bypassed by small-value mylar types to maintain the transient response ((desirable as large electrolytics have a degree of self-inductance) while the two stages are directly wired to each other and the cartridge without any coupling capacitors.

The power amplifiers are designed to have a high-current capability and, as a result, it has not been necessary to fit any protection circuits (other than the mains fuse required for safety and fire prevention). The circuitry is fairly conventional but the voltage gain stages are all fed from a stable power supply completely isolated from the high-current supply feeding the output stage. This arrangement avoids the usual problem where the performance of the gain-stages varies depending on whether the amplifier is playing loud or soft.

TEST RESULTS

The initial impressions on audition were very good and it was clear that these products were well above average in terms of their sound quality. The sound was extremely clear and, after switching from a well regarded British pre/power combination, the effect was akin to opening the curtains which previously hung in front of the speakers. Much detail was revealed and yet there was none of the artificial brightness that normally accompanies such openness, for tonally the PS was quite flat with a slight (but very slight) mid-range emphasis. The sound had a nice solidity with precise stereo imaging which only collapsed at high listening levels. The main weaknesses were in the power amplifier which could benefit from more power and an improvement in its high-frequency smoothness to remove a slight raggedness.

This PS combination only just achieved its rated power output with both channels driven. The power output rose nicely with reducing load impedance (95W into 4ohms, 130W into 2ohms) showing plenty of output current capability.

The harmonic distortion was rather poor at 40W and showed a lot of crossover artifacts.

The preamplifier results were quite reasonable although the disc amplifier was not amongst the quietest tested. The RIAA equalisation was very flat and accurate although the bass began to roll-off rather early.

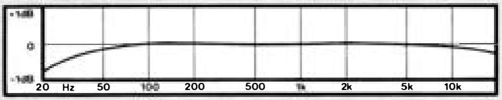
THE VERDICT

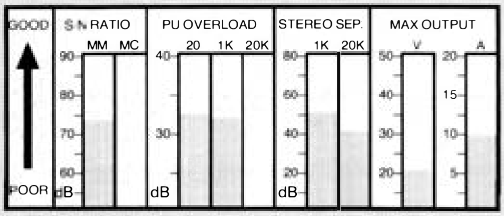
This PS system will cost you about £560 in the shops and therefore can't be said to offer particularly good value for money especially when the standard of finish is considered. But there is no getting away from the fact that in terms of fulfilling its function the PS Audio models do an excellent job. The sound quality put many of our better regarded British products to shame and restored my interest in a lot of old recordings which I had but all discarded. Be warned — it can sound different to some of those amplifiers you are told are good, but take some time over listening and you'll find that 'different' doesn't always mean worse.

TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	51	52	52
(WATTS)	8+8Ω	41	42	42
	4Ω	91	95	95
HARMONIC DISTORTION (% @ 40 WATTS)		0·09	0·27	0·3

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	3Hz	30kHz





PIONEER A5

One of the smaller of Pioneer's unusually styled amplifiers, the A5 offers a basic set of facilities and some 35W per channel. When this range was first announced their appearance generated much comment but they have found acceptance with the high street buyer. The dark central panel houses a number of indicators which are intended to gave an 'at-a-glance' listing of the various switch settings. Unfortunately I found it to be initially confusing which must be bad news to the behavioural scientists at Pioneer, but familiarity does help. Still there was no doubt about the standard of finish partly achieved by the use of a high-quality plastic moulding for the fascia. The vertical row of switches select the three normal and two tape inputs while a rotary switch selects the three normal and two tape inputs while a rotary switch selects the tape output source but only permits dubbing from Tape 1 to Tape 2. The tone controls can be switched out of circuit but there are no filters only the usual loudness switch. There are also switches for two pairs of speakers and a mute (−20dB) button.

CONSTRUCTION & TECHNOLOGY

Open up the Pioneer A5 and surprise, surprise there's nothing there. Well, not nothing exactly, but there is an awful lot of empty space. The PCBs are also used in the more expensive and complex models so in the A5 they seem to be full of holes and large blank areas where components aren't. The chassis layout is fairly tidy, being split by a large U-shaped aluminium heatsink, but loading of the PCBs is rather messy. However, the components used are of good quality and the wiring, while a little untidy, is cleanly connected using wirewrapping posts.

No circuit schematic was provided but the disc pickup stage proved to be a simple arrangement using a dual integrated circuit amplifier. The power-amplifiers have a fully complementary output stage with the tone controls wired into its negative feedback loop. The power supply is common to the two channels and DC offset voltage protection as well as a delayed speaker switch-on is provided by a relay and its sensing circuits.

£140 MERIT ★★★☆☆
VALUE ★★★☆☆

TEST RESULTS

The A5 developed a generous output power into 8ohms, the single channel figure at 68W being almost twice the manufacturer's rating. However, the power into 4ohms was restricted by the protection limiters, although a healthy 84W was managed (1kHz). Into 2ohms though power fell back to 36W and the protection occasionally tripped out.

The harmonic distortion was low at mid-band frequencies and rose a little at higher frequencies but not enough to be of any significance. The frequency response was fairly sensibly curtailed albeit rather excessively so at the top-end, the −1dB points being at about 8Hz and 18kHz.

The input stage measurements gave generally good results but the stereo separation was rather disappointing particularly at 20kHz. The RIAA equalisation proved to be quite accurate in the midband showing a gradual roll-off at each end of the audio band.

The listening tests showed the A5 to be quite a nice little amplifier. The midband was particularly pleasant with fairly precise imaging with a nice sensation of depth and width. The sound had a clear and open quality and fairly detailed although a certain 'smoothness' could also be said to have resulted in less precision. The bass was thought to be reasonably firm but the treble was rather muddled and inaccurate. Overall the sound was smooth and pleasant but seemed comparatively lifeless with some pieces of music.

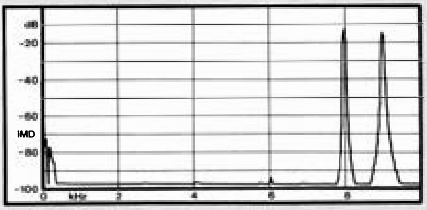
THE VERDICT

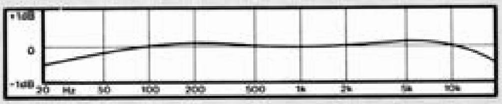
For a 35W amplifier with no moving-coil input the A5 is a rather expensive purchase. However its styling makes it visually more interesting — and interest always costs a little extra! A competent enough design which produces an inoffensive if unexciting sound.

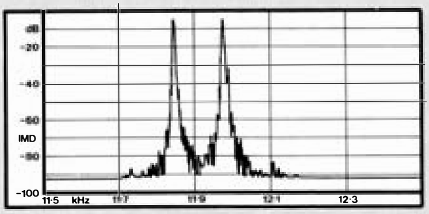
But it is something of a safe purchase in the jungle and that's worth a weight off anyone's mind.

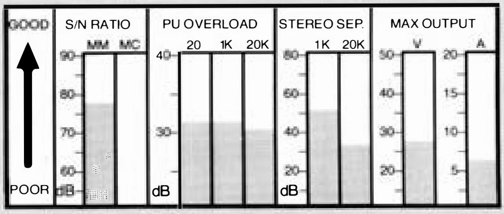
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	65	68	66
(WATTS)	8+8Ω	52	57	55
	4Ω	81	84	81
HARMONIC DISTORTION (% @ 35 WATTS)		0·010	0·007	0·030
FREQUENCY RESPONSE		LOW		HIGH
BETWEEN −1dB POINTS		7Hz		18kHz









PIONEER A7

The A7 is the 70W model in Pioneer's line up and at first glance is deceptively simple with just a row of selector switches volume, bass and treble rotary controls and the large central 'National Grid' display. A selection of arrows, dashes and strange international symbols are used to clearly identify the route the signal will be taking through the amplifier. So far, so good, but the only trouble with these international symbols is that you don't recognise them whatever your nationality! Oh yes, and Pioneer couldn't resist the temptation to slip a couple of thermometer scale power-meters into this display.

After a time you discover that the left hand side of the fascia hinges down and slides underneath out of harms way, to reveal a number of extra controls including switching for moving-magnet and moving-coil sources; loudness and subsonic filters (only 6dB/octave) and switching for the two pairs of speakers. There is also a bypass switch for the tone controls and a tape source switch which allows tape dubbing to be set up in either direction between the two top connections.

With its light gold finish the overall appearance of the A7 is smart, although when everything is pressed the array of lights can be distracting.

CONSTRUCTION & TECHNOLOGY

The internal construction of the A7 is neat and business-like, if rather complicated. The circuitry is provided by some 15 PCBs linked by either board-to-board connectors or cable-looms. Good quality components are used and although the loading of some of the boards is rather messy, it is obvious that a lot of money has been spent on parts.

A common transformer (of good size) and power supply is used for both channels.

No circuits were provided but as far as I could tell the disc amplifier is discrete and uses a differential configuration with a pair of low-noise FETs at the input. The RIAA equalisation is active using a network in the negative feedback loop. Virtually the same circuit is used for moving-coil cartridges but the gain and input loading is switched by a miniature relay on the board.

The tone-controls are wired around an IC amplifier stage but the power amplifier is discrete, again using a complementary pair of large power transistors. Two bias networks are used: one fixed; one floating to ensure that the output transistors are never in a non-conducting state.

£250 **MERIT** ★★★★☆ **VALUE** ★★★★☆

TEST RESULTS

The power output into 8ohms generously exceeded the manufacturer's rating, nor was there too much difference between the single and both channel figures. The power rose nicely to 180W driving a 4ohm load and, for short periods, to 230W into 2ohms after which the protection circuits screamed enough! So a good result and difficult loudspeaker loads are unlikely to be a problem.

The harmonic distortion was virtually negligible and those IM sidebands which show on the spectral plots are fairly low in level. The preamplifier section also returned good results, the noise being particularly low, although once again as with the other Pioneer models the frequency response of the A7 is rather unnecessarily extended.

The RIAA response falls within a 0·3dB wide band — a good result — and generally shows a gentle slope down at either end.

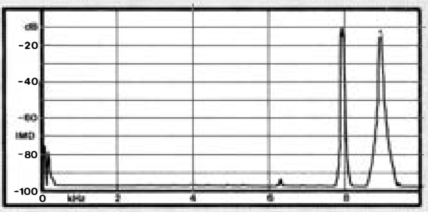
With some music this Pioneer really excelled whereas other programme material highlighted its weaknesses. Midband frequencies were handled particularly well, the sound being clear, open and detailed, solo female voice being reproduced with a nice naturalness. The problems came at the frequency extremes; the bass while being fairly well controlled seemed to lack real power especially on such transients as tympani being struck. The treble end suffered from a slightly metallic tone and lacked smoothness but this wasn't to a degree that was unpleasant. Overall a good sound.

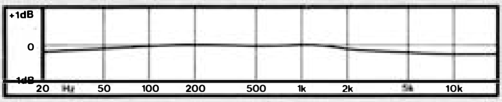
THE VERDICT

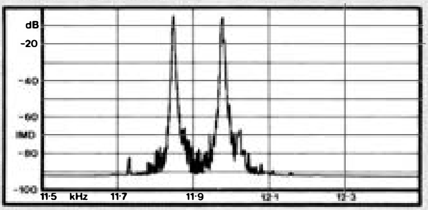
The A7 is a nice amplifier which did well in both the laboratory and the listening tests. Compared to some of its competitors it might seem expensive but it could well justify a 90W rating to make it particularly good value in view of its interesting design and good sound quality.

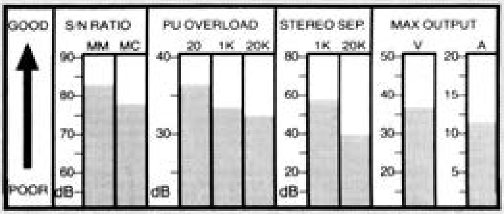
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	110	118	115
(WATTS)	8+8Ω	94	104	104
	4Ω	162	182	175
HARMONIC DISTORTION (% @ 70 WATTS)		0·005	0·003	0·007
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 3Hz	HIGH 92kHz	









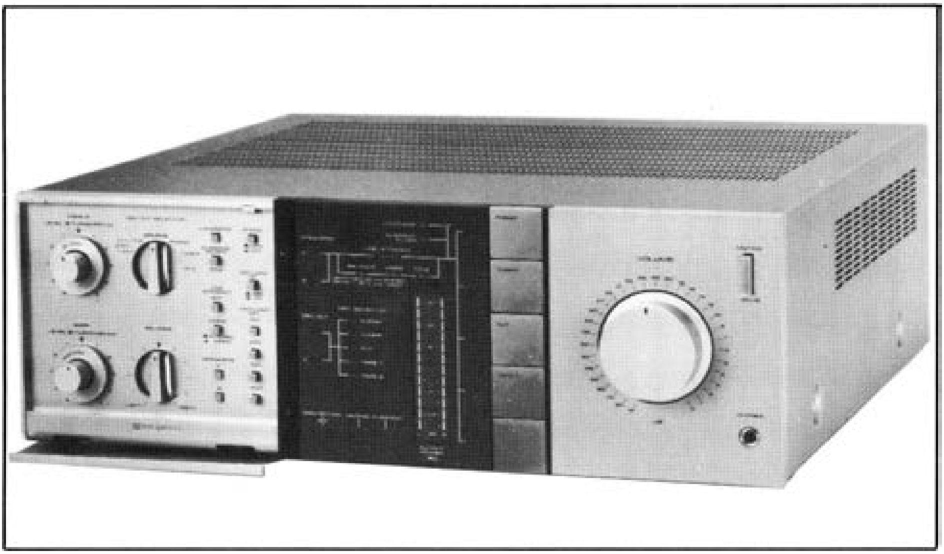
PIONEER A9

The A9 is the biggest of Pioneer's integrated models. It is also very expensive although the 34lb weight of this 100W model at least suggests that the owner is getting plenty of hardware for his money.

Like the other Pioneer 'A' series the A9 has apparently been styled by an engineer from the CEGB. The central display is obviously intended to clearly identify the more important control settings as well as housing a brightly-lit power meter display. However I found this amplifier rather difficult to use the combination of many poorly identified switches and the display leading to as much confusion as clarification.

The A9 has three inputs (the disc input switchable between moving-magnet and moving-coil) and two tape connections with dubbing possible in both directions. The left hand side of the fascia which ostensibly only carries the bass and treble tone controls does in fact hinge open to reveal the balance and tape-monitoring controls as well as minor switches for mode, tone-cancel, loudness and subsonic filters and speakers A and B. Other small switches determine the input loading of the disc stage either 33 or 100ohms on moving coil and 100/200/300/400pF in parallel with 50kohms for moving-magnet.

The A9 also has a rather unusual feature a sort of subsonic frequency detector which operates yet another indicator on the display panel. The idea is that whenever the indicator is illuminated the subsonic filter should be operated.



CONSTRUCTION & TECHNOLOGY

The chassis of the A9 is extremely crowded and it is hard to see how Pioneer could find space for another transistor. A total of 15 PCBs are joined together by cable looms with a mix of wire-wrap and plug/socket connections.

The single transformer feeding the common power supply is a large toroidal type and the reservoir capacitors are housed in a screening box to prevent the charging spikes being induced into the low-level circuitry (necessary with such a crowded chassis).

The circuitry uses a mixture of ICs and discrete transistors. The moving-coil stage is discrete using a complementary array whilst the main disc stage uses an IC with a dual low-noise FET at the front end. The tone control amplifier is based around another IC.

The power amplifier has two parallel pairs of complementary power-transistors and the Pioneer 'Non Switching' dual bias networks (one floating; one fixed) to ensure that the output transistors are never in a non-conducting state.

£450 **MERIT** ★★★★☆
VALUE ★★★★☆

TEST RESULTS

The A9 delivered plenty of power into the 8ohm load and there was little difference between single and both channel operation this suggesting a stiff power supply. The power output fell slightly to 163W into 4ohms and 80W into 2ohms, these figures being determined by the operation of the protection current limiter. The peak current measurements showed that only 9 amp peaks could be sustained equivalent to 6·3 amps continuous RMS.

The harmonic distortion was low but showing a rise at high frequencies and there were some minor sidebands in the IM spectral plots. The frequency response was thought to be unnecessarily extended at both ends, although the subsonic filter could be used.

Otherwise the preamplifer stages returned good results, both moving-magnet and moving-coil inputs having low noise, good overload margins and fine 20kHz separation.

As far as sound quality is concerned there wasn't a great deal to be found wrong with the A9. Like the A7 the weaknesses are at the frequency extremes and the same characteristics were found (see A7 report previous page). But despite having a high power output on paper the A9 wasn't able to reproduce music at the high levels expected; the treble in particular exhibiting both a hardness and metallic tone. However, used at 'sensible' levels the sound was clear with good revelation of low-level detail and good handling of the more dynamic passages.

Despite its rather flashy Japanese image this amplifier did well in the listening tests. The sound quality was clear and open with good revelation of detail. Both treble and bass were controlled, the bass in particular being very solid, powerful and going down a long way. The sound remained accurate at high levels but at all levels the imaging was diffuse particularly in terms of front-to-back depth

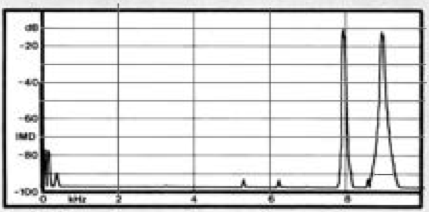
THE VERDICT

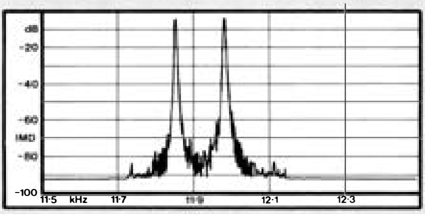
The A9 achieved good results for its measured performance and also did well in the listening tests. It is a well equipped and powerful amplifier which undoubtedly holds some appeal to the technology freaks amongst us. Although it is rather expensive it is in fact no more than several well regarded audiophile combinations which sound no better.

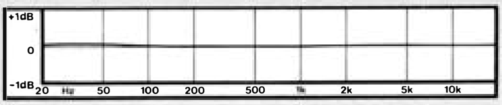
TEST RESULTS

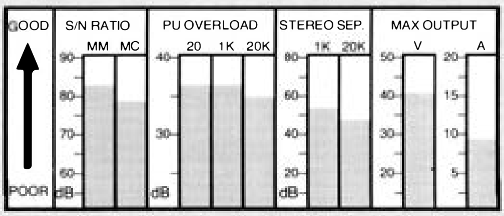
TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	175	180	180
	8+8Ω	170	178	175
	4Ω	163	163	163
HARMONIC DISTORTION (% @ 110 WATTS)		0·005	0·002	0·061

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	3Hz	91kHz









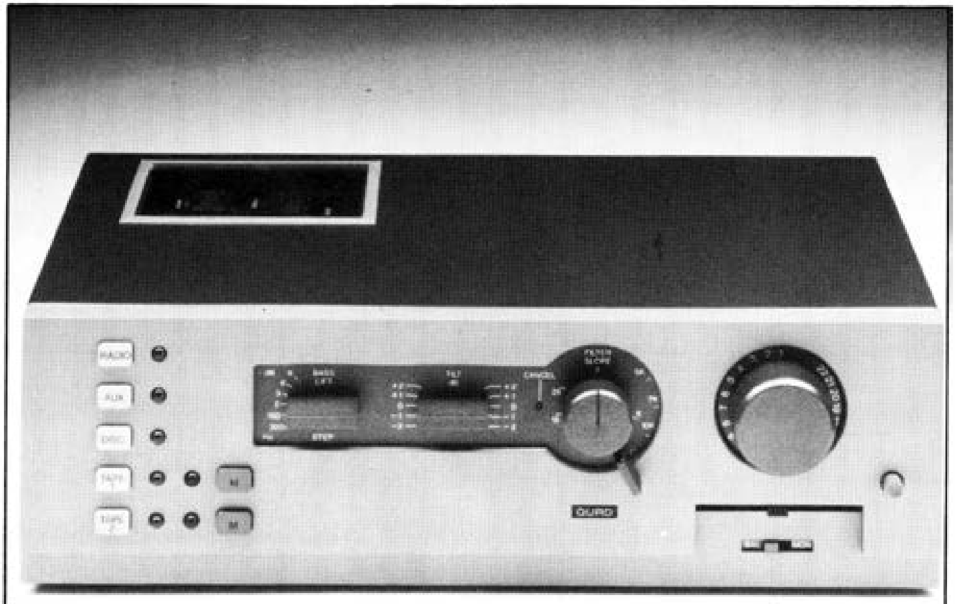
QUAD 44 405

This amplifier/preamplifer combiantion has been around a fair few years now but, in the best Quad tradition can be expected to be around in gradually refined versions for many more to come. Both units are neatly styled in the rather idiosyncratic Quad manner and the preamplifier has an interesting array of controls which are certain to confuse those brought up on a succession of Japanese amplifiers. The five input selector switches are light-action 'touch' types, the operation of which lights an adjacent LED. Two further switches permit monitoring of the two tape sources. The large volume control has 22 detented positions giving accurate level setting (in 2dB steps over part of its range) and good tracking between channels.

The slider balance control has a number of functions for when the 44 is switched to mono it can be used to mix the two channels or can feed either source channel to both outputs. A well thought-out and useful arrangement. Unlike the earlier 33 preamplifier the 44 has no conventional tone controls but does retain the familiar HF filter having three turnover frequencies and completely variable slope (0 to 25dB/octave). The oddly named 'tilt' control does just that — it gives equal and opposite cut and boost to the high and low frequency extremes whilst leaving the midband level unchanged. Finally the bass control offers a choice of several degrees of cut and boost.

A variety of user-changeable input and output modules can be supplied, amongst them input amplifiers for both moving-magnet and moving-coil cartridges and for microphones. Easily accessible switches can be used to vary the input loading and sensitivity of the disc stage and the input and output levels of the tape sections.

By contrast the 100W (or more correctly 200VA) amplifier just has a 4 pin DIN input connector and one pair of speaker terminals with a power LED set into the front panel. But it does incorporate Quad's highly innovative current dumping circuitry.



Quad 44

£270 MERIT ★★★★☆
£299 VALUE ★★★☆☆

CONSTRUCTION & TECHNOLOGY

The internal construction of the 44 is very neat and well thought-out. Typical of the standards adopted is the power supply which is totally enclosed by a red plastic box sufficient to obstruct any prying fingers. All the input and output circuitry is on plug-in modules which all attach to a long mother board. These are clean and tidy although not marked with component designations.

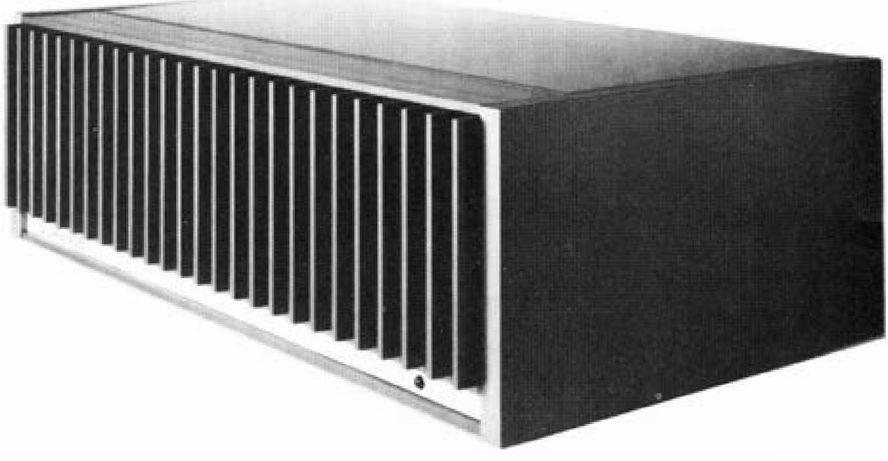
The layout of the 405 is about as simple as a power amplifier could be. At either end of the chassis there is a power-board bolted onto the heatsink. In between there is a large encapsulated power-transformer and associated rectifier and capacitors, the single supply feeding both channels.

Both products use some novel circuit techniques. The 44 preamplifier uses CMOS electronic switches to route all the input and output signals onto common busbars. Both the moving-magnet and moving-coil disc amplifiers use the same type of complementary common emitter input stage feeding an inverting operational amplifier with overall negative feedback around the two stages.

The RIAA equalisation has the low-frequency time-constants in the feedback loop while the high-frequency section is made up of a passive filter at the output.

The 405 uses the current dumping concept where the amplifier consists of two sections; a low-power Class A amplifier of excellent linearity and a high power Class B output stage — the dumpers of current. Both drive the load but at low signal levels the loudspeaker signal comes from the Class A stage than at higher levels the current dumpers turn-on to deliver the bulk of the current.

The clever bit is the arrangement so that feedforward and feedback components are in balance such that the quality of the output signal is wholly dependent upon the performance of the Class A stage. This balance is achieved by matching four pasive components thereby avoiding the need for any preset adjustments.



Quad 405

TEST RESULTS

With both channels driven the 8ohm power was marginally above the manufacturer's rated power but current limiting restricted the 4ohm power to about 95W and the 2ohm power to 46W. The peak instantaneous current was restricted by the current limiter to 6amps so obviously the 405 is not suited to the more 'difficult' loudspeaker loads.

The harmonic and IM distortions were both very low although the former rose quickly above 20kHz causing the effective power bandwidth to be restricted.

The 44 preamplifier returned good results with the bulk of the distortion occurring in the electronic switch circuits. The stereo separation was high, particularly the 20kHz figure which puts many, more expensive, units to shame. One weak area was the way in which the input stages clipped; some slew-rate limiting being visible (eg triangulation of the sinewave) However the headroom is quite adequate in practice, the low 20kHz figure being a result of using passive HF equalisation

The Quad 44 preamplifier generated little adverse comment on test the sound quality being quite acceptable. In comparison with a couple of more expensive models there was an evident loss of some low-level detail and a lack of overall 'transparency' of the sound stage but at its price it is one of the better preamplifiers.

By contrast the performance of the 405 power amplifier seemed to be very speaker dependent (as might be expected from the measurements). Problem areas were a lack of control of the low-bass; and a tendency towards confusion and muddle in loud choral passages. But with a careful choice of loudspeaker the results were quite satisfying. With limiter (unofficially) disabled the performance was much improved and, in loud passages, a firm controlled bass was heard.

THE VERDICT

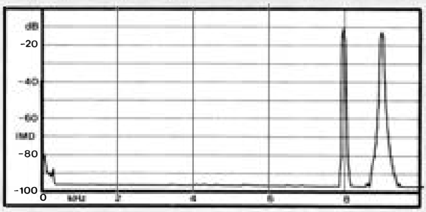
In some ways Quad's own reputation is its biggest sales aid so our comments are unlikely to have much influence. But the Quad 44 did prove to offer a good all-round performance; a range of useful facilities free of gimmicks; and good value for money — a product worthy of recommendation.

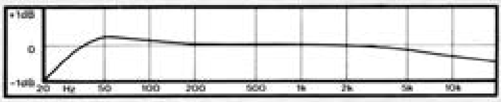
The power amplifier is only capable of driving 'easier' loudspeaker loads and if paired with, say, Linn Isobariks its inadequacy should not be unexpected. But used with the right loudspeakers it gives good results.

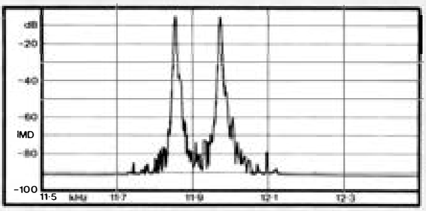
TEST RESULTS

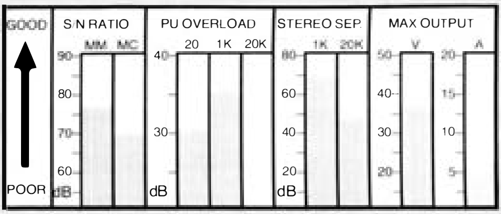
TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	107	111	109
(WATTS)	8+8Ω	103	105	103
	4Ω	87	95	90
HARMONIC DISTORTION (% @ 100 WATTS)		0 01	0 006	0 003

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	15hZ	20kHz









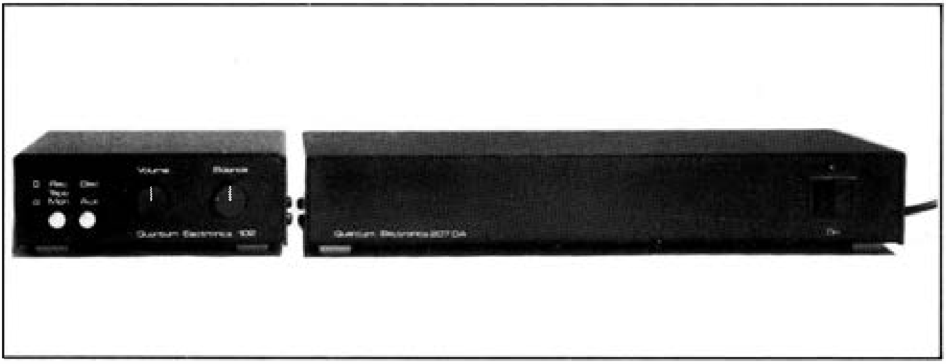
QUANTUM 102 2070A

The Midlands-based company of Quantum Electronics has, over the past few years, built up a reputation for their do-it-yourself amplifier kits. They have now gone 'legitimate' and become true manufacturers with quite an impressive range of sensibly priced pre and power amplifiers. For this particular review we looked at their 102 preamplifier and the type 2070A stereo power amplifier which is rated at 77W. The styling is typically audiophile which means there isn't any! Instead the circuitry is housed in fairly utilitarian simple steel boxes designed for price rather than looks although, that said, the standard of finish was found to be very high with good quality black paintwork.

The 102 preamplifier is powered (via a DIN-to-DIN umbilical) from the power amplifier and offers only the more basic facilities. The two rotary controls are for volume and balance whilst two push-buttons select line or disc input and tape-monitor. Inside the box there is access to a switchable LF filter and a range of disc input boards may be fitted to match either moving-magnet or moving-coil cartridges and offer a choice of input sensitivity

Most of the rear panel connections are via DIN sockets but (presumably for user convenience) the disc input uses a pair of phono jacks.

The power amplifier has a mains power switch and indicator on the front panel, but at the rear, together with the DIN input and banana output sockets, there is a switch which selects the input signal for stereo (L & R) or mono (L or R) operation; the latter for when it is used as part of an active system.



CONSTRUCTION & TECHNOLOGY

Access to the Quantum amplifiers is difficult for the average user because the cases are secured by socket headed screws.

The 102 preamplifier is housed in a simple aluminium box and has all the circuitry assembled onto a single board with no wiring at all. The components used are of generally good quality with 2% capacitors in the RIAA networks, although some eyebrows may be raised over the use of Korean transistors! On board switches are provided for the subsonic filter; for aux. input sensitivity (150mV or 775mV) and tape monitor sensitivity. A small plug-in board carries the alternative input loading components for the disc amplifier.

No circuit details were provided but a mixture of low-noise ICs and transistors are used. In all there are four ICs and 26 transistors, so there has been no skimping on parts.

The internal construction of the power amplifier is also to a high standard, the chassis being dominated by the large toroidal power transformer. Although this transformer is common, both channels have their own rectifiers and reservoir capacitors.

The power boards, bolted to rather undersize heatsinks are nicely built with adequate fusing. A quasi-complementary output stage is used, the NPN output transistors being the well regarded Toshiba 2SD424 types. One interesting point is Quantum's use of a coiled length of resistance wire used in place of conventional emitter resistors.

£92 MERIT ★★★☆☆
£195 VALUE ★★★★☆

TEST RESULTS

The 2070A proved to be a powerful enough amplifier at low and midband frequencies although there was some loss of output due to the collapse of the power supply at low frequencies. Thus into 8ohms we recorded 104W and 92W then into 4ohms 132W and 150W for 1kHz and 20Hz respectively. However the measurements at 20kHz were plagued by the fuses blowing. Eventually we recorded 80W (8ohm) and 110W (4ohm), at which powers the distortion shot up and the fuse blew.

Midband distortion was no problem and the overall bandwidth was well controlled but the damping factor steadily declined with rising frequency reaching a fairly low 30 at 20kHz.

The preamplifier proved to be fairly noisy in use but returned good overload figures. The RIAA equalisation was accurate within a 0·4dB band from 50Hz but below that the bass was steadily rolled-off by the internal subsonic filter to be down 3·3dB at 20Hz. Stereo separation, while very good at 20kHz, was a little low at 1kHz.

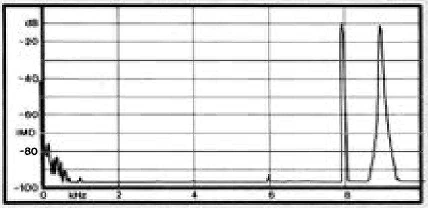
This Quantum combination did fairly well in the listening tests producing a very acceptable sound quality. The detail was well preserved and the midrange was clear, albeit a little forward in character. The bass was well defined but control was loose at high levels. The treble lacked precision, this being particularly noticeable in complex passages but overall the sound hung together well.

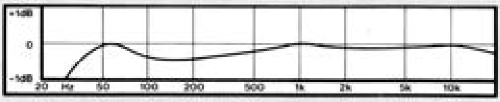
THE VERDICT

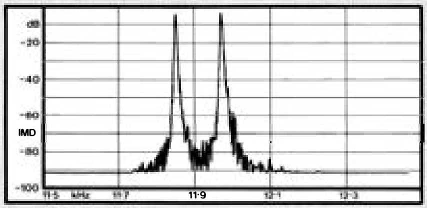
The chunky knobs apart this is probably the smartest and best built of the British 'cottage industry' products. It is also a very attractive buy at the low price quoted. In the course of our tests the power amplifier seemed reluctant to deliver a lot of current at high frequencies but otherwise seemed to be a competent enough design. The sound quality was free of any irritating vices and was most pleasant. Overall this combination is worthy of recommendation.

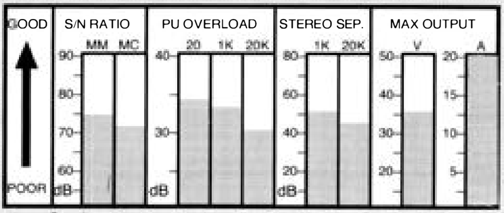
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	92	104	80
(WATTS)	8+8Ω	81	82	75
	4Ω	182	210	163
HARMONIC DISTORTION (% @ 70 WATTS)		0·061	0·009	0·03
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 30Hz		HIGH 25kHz

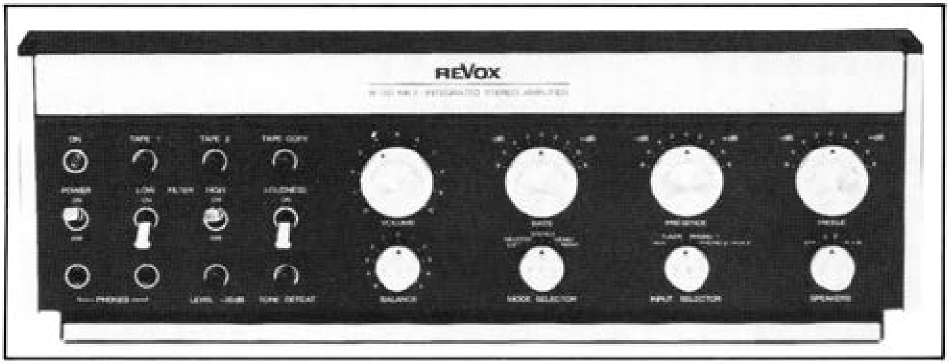








REVOX B750-II



The B750 is a big, heavy amplifier built in the Revox engineering tradition although only rated at a moderate 75W per channel. The styling is distinctive and perhaps rather too 'technical' for many people's tastes but there is no doubt that when used with its companion B77 or A700 tape recorders the overall effect is very smart. The finish is in dark blue and grey with silver control knobs and the front panel, while neatly laid out, takes some getting used to as the controls always seem to be in the wrong place. This is all the more confusing because the Revox has an excellent array of facilities including three inputs (but not one for moving-coil inputs) and two tape connections with cross-dubbing. Switches control mode, loudspeaker selection, muting (– 20dB), high and low filters and loudness. Unusually the B750 has two headphone sockets and, under the hinged top part of the panel; a record-out jack socket making easy connection to a temporary tape-deck installation. This hinged flap also conceals a pre-power break switch making possible separate operation of the preamp and power-amp sections, a disc sensitivity control and a disc input impedance switch (25/50/100kohms).

The B750 has the usual bass and treble controls but also has a presence control centred at 3kHz and so affecting the midband frequencies. All these controls are disabled by operation of the tone-defeat toggle switch.

Around the rear the Revox uses a mix of phono and DIN sockets which are in a recessed section permitting flush wall mounting — fine for permanent installations but inconvienient if you are the sort who is always connecting new equipment.

CONSTRUCTION & TECHNOLOGY

Once the lid is taken off (simply by removing two screws) the A750 is a joy to behold. The standard of construction is just what you'd expect from Revox. Most of the circuitry is readily accessible for service and most of the interconnections are made through the use of plug-in cables and board links. All the circuit boards are neatly laid out with good quality components (virtually all the electrolytic capacitors are the expensive tantalum types) but the board surfaces are, regrettably, not screen-printed with component designations. The circuitry is housed in a strong steel case which has extensive screening plates between different circuit areas and this results in a rigid assembly.

A large efficient C-core power transformer is used and has separate secondary windings which feed a power supply for each channel.

No circuit information was available at the time of writing but it was obvous that much of the amplifier circuitry used discrete components and not ICs (this being a change from some previous Revox designs). The circuitry appeared conventional with the disc-stage using RIAA equalisation arund a series feedback loop and with every input passed through a unity-gain buffer amplifier to prevent any loading problems with ancillary equipment. The tone-controls use rotary switches instead of the cheaper detented pots and, in the case of the B750, these are connected to laser-trimmed resistor networks designed to ensure the accuracy of the control setting within a fraction of a dB.

The power amplifier is fully-complementary and appears to use a mirror-imaged circuit configuration with a pair of large metal-can power transistors.

£552 MERIT ★★★☆☆
VALUE ★★★☆☆

TEST RESULTS

Power delivery was good into all the resistive loads with a useful rise to 160W into 4ohms. However, the power into 2ohms fell to a mere 30W following operation of the protection circuits, although the burst power was nearly 300W. It proved capable of a high peak current (some 20 amps at saturation) but the Revox was troubled by the capacitive load which seemed to lead to current limiting. Harmonic and IM distortions were low but not especially so.

The overall bandwidth was rather extended at high frequencies but the equalised response of the disc input was accurate across the midband (within about 0·2dB limits), although the errors became significant at the frequency extremes. The bass cut shown suggests that a subsonic filter may be incorporaed in the disc amplifier. The rising response (+0·6dB at 30kHz) would also suggest an audible brightness (this rising response continues, reaching +5·9dB at 90kHz). The other measurements were fine with fairly low-noise levels on disc and good input overload capability. The stereo separation was also adequate although it fell to 33dB at 20kHz.

On audition the Revox displayed a degree of brightness which was acceptable with the Rogers Studio One speakers giving a sense of clarity, but less successful with other speakers including the Quads.

The sound quality was clean, open, indeed almost analytical and the bright character was perhaps emphasised by a bass lightness; but such bass as was heard was well controlled and detailed.

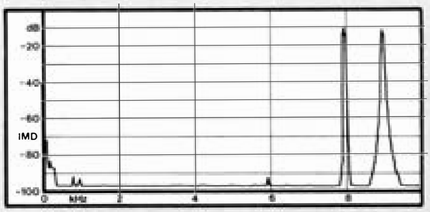
With those loudspeakers with a falling response, the Revox could be the basis of a viable system but with many speakers the overall sound, although exciting and dynamic in small doses, was eventually fatiguing.

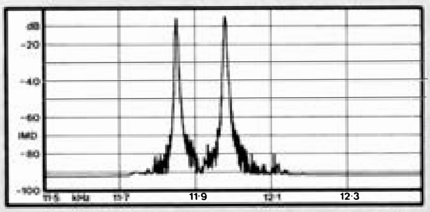
THE VERDICT

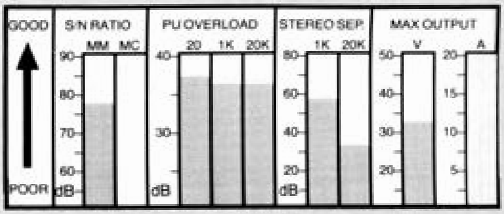
The Revox B750 is an expensive amplifier, well built, and gives a moderately good account of itself. Because of its price it is unlikely to appeal to many enthusiasts as a single purchase but will be seen as a 'must' by many Revox tape-recorder owners. They are unlikely to be dissatisfied for the B750 has a clear, open sound quality. Care is needed in the choice of loudspeaker (so listen before you buy) and a little bass-boost was found to give a better tonal balance. I would also add that my own personal experience with Revox products suggests excellent consistency and reliability.

TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	100	106	105
	8+8Ω	96	105	94
	4Ω	147	160	152
HARMONIC DISTORTION (% @ 75 WATTS)		0·082	0·074	0·12
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW	HIGH	
		8Hz	75kHz	







ROGERS A100

The A100 is the higher powered of Rogers' range of two amplifiers (the other being the A75) and is rated at 55W per channel. The styling is quite restrained with a black front panel and knobs finished to a high standard and nicely set-off by the veneered side panels. The overall effect is rather spoiled by the shiny black top-cover although this will be out of sight when the A100 is shelf mounted. Three inputs are provided including a moving-magnet disc input whose loading can be changed to give 47k or 100kohms and another back panel switch to add 60pF or 300pF of shunt capacitance. Connections are provided for two tape-recorders, but dubbing is only possible in one direction. One unusual feature is a DIN socket on the front panel which is the auxiliary input but also has an output feed tapped just before the volume control. It is, therefore, a convenient means of quickly connecting a cassette deck temporarily without having to poke around the often inaccessible back-panel cabling. Bass and treble controls are provided and these (and the high filter) can be bypassed to give a flat response. The high frequency filter is quite unusual, being rather like that in the Quad control unit. Two turnover frequencies (6kHz or 9kHz) and a slope control vary the rate of attenuation from zero (ie, filter is inoperative) to 18dB/octave.

Two pairs of loudspeaker can be connected and can be individually switched. Finally the A100 has three shuttered mains supply outlets for ancillary equipment with the supply switched by the amplifier's power switch. Also noteworthy was the comprehensive and informative manual provided with this model.

CONSTRUCTION & TECHNOLOGY

The first problem with the A100 was actually getting inside the thing, for it is extremely solidly built in the British 'Dreadnought' tradition. The main preamplifier board runs along the case side from the input sockets to the selector switches and is fully screened by steel panels. A second board sits behind the front panel and carries all the tone control and filter circuitry. The two power amplifier boards are each bolted to the small extruded heatsinks which carry the output transistors. A single power supply is used for both channels.

All the boards are glass fibre, screen printed with circuit designations and loaded with good quality components. The wiring and soldering is, in general, neatly done.

No circuit information was provided by Swisstone so a few educated guesses had to be made. The preamplifier uses discrete transistors and the disc amplifier in particular seems to be quite complex using eight transistors per channel with a differential input stage, an active load and a push-pull constant current output stage. RIAA equalisation is achieved in the negative feedback loop of this stage.

The power amplifiers are relatively simple, being AC coupled and using an IC to provide the voltage gain and a quasi-complementary output stage to provide current gain.

No voltage selector is fitted but the mains transformer has 120 volt taps to permit conversion.

TEST RESULTS

The A100 maintained its output well into different load impedances; the single channel power-output being 76W, 115W and 109W into 8, 4 and 2ohms respectively. Loading with the partly capacitive network also had very little effect upon the Rogers' maximum undistorted output, so this amplifier would seem to be able to cope with most loudspeaker loads.

Peak current capability was high at 17·5 amps and the test revealed that the protection circuits were not prematurely limiting the available drive to the loudspeaker.

Harmonic distortion was a little higher than expected, but still within the manufacturer's specification.

Turning to the preamplifier section, the stereo separation was fine as was the overload margin on disc. The noise levels were rather high, though, on both disc and tuner inputs. The equalised response through the disc input closely followed the RIAA response except at low frequencies where a degree of bass roll-off occured because the A100 follows the generally ignored IEC response recommendations.

On audition the A100 was rated well above average, the tonal response seeming to be very flat. The sound was clear without being excessively bright and a fair amount of detail was heard, particularly at low frequencies which were reproduced well. At high levels the very slight treble emphasis became pronounced and irritating with some loudspeakers, but partnered with the Rogers Studio Ones the balance seemed more neutral and the overall sound was thought to be smooth yet clear.

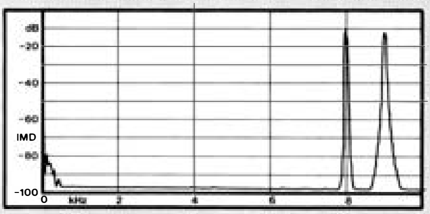
THE VERDICT

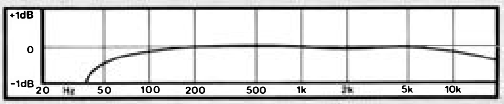
In many respects the Rogers A100 has always been an under-rated model for this manufacturer's marketing emphasis has always concentrated on their loudspeakers. In our tests it gave a very creditable performance and its sound quality was good enough for it to be 'borrowed' for my second domestic system for some months.

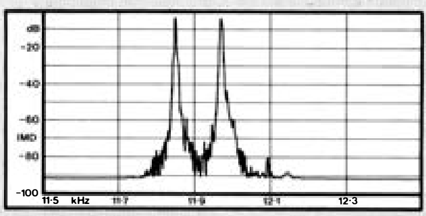
The price is rather high in relation to the product's current modest image but it is a worthwhile investment if you seek a competently designed amplifier that is capable of bringing much pleasure when listening to your music.

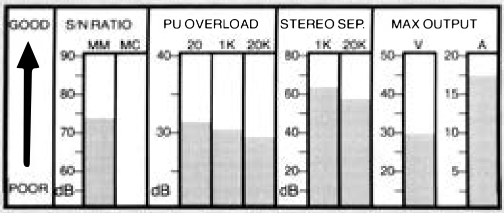
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	72	76	73
(WATTS)	8+8Ω	60	65	63
	4Ω	109	115	113
HARMONIC DISTORTION (% @ 55 WATTS)		0·036	0·042	0·033
FREQUENCY RESPONSE		LOW	HIGH	
BETWEEN −1dB POINTS		28Hz	33kHz	









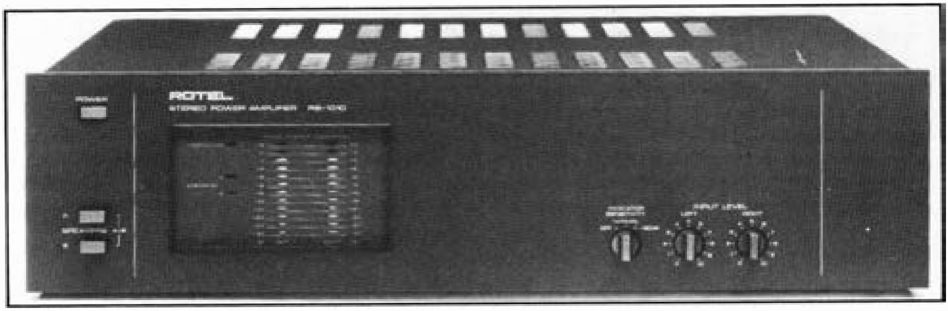
ROTEL RB1010 RC1010

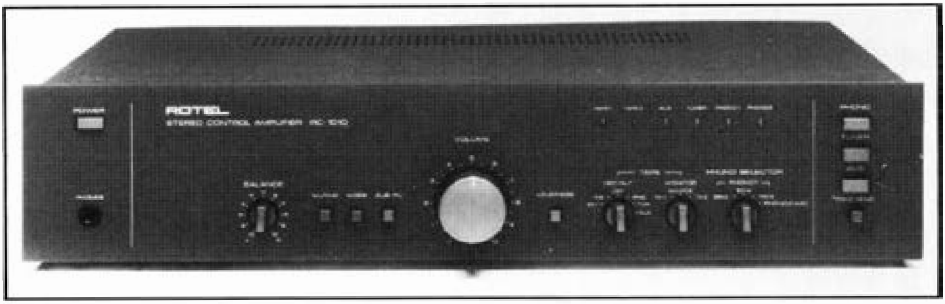
The 1010 pre and power amplifiers are currently top of the Rotel product range and, in common with the rest of their models, this 100W system seeks to offer unusually good value for money. At a combined price of around £300, this could be well the case.

The power amplifier has switching for two pairs of loudspeakers plus the ubiquitous LED power meter with extra lights showing the operation of the speaker switches and the protection relay. The display range can be decreased, or the display switched off, by a front panel knob, and there is a separate gain control for each channel.

The matching RC 1010 preamplifier has inputs for both MM and MC cartridges with switched input loads for the former. Two tape decks can be connected and there is some very flexible tape-switching provided which, amongst other things, allows dubbing in both directions. The RC-1010 is a straight-line model so there are no tone controls but it is possible, by pressing a front-panel switch, to patch in an external equaliser.

The styling gives these units a typically Japanese rack system appearance with their plain flat front panels and attractive black/grey finish.





CONSTRUCTION & TECHNOLOGY

Both units are well constructed and use good quality components. The RB-1010 has virtually all its circuitry on one large PCB mounted between two heatsinks. Each channel has its own power supply fed from a large common toroidal transformer. The preamplifier makes much use of remote cable operated switches to cut down on the amount of signal wiring, but even so there are a lot of interconnections which give the chassis an untidy appearance.

The power amplifier circuit is a conventional DC-coupled design with much use of cascode stages to improve the linearity and bandwidth of the voltage amplifiers The output stage is fully complementary using two parallel pairs of high speed 250W transistors.

This Rotel incorporates two interesting design features. The first is called a DC Servo. Only a moderate amount of negative feedback is used around the power amplifier and this is insufficient to stabilise the DC voltage drifts which occur with temperature changes. As a result there could be a

£175 MERIT ★★★☆☆
£126 VALUE ★★★★★

large and varying DC offset voltage across the loudspeaker. So Rotel have a second feedback loop where the AC signal is filtered out and the residual DC is amplified by an operational amplifier to give an error signal to feed back to the input stage. This 'active' system increases the DC negative feedback by some 40dB but has no effect on the performance at audio frequencies.

The RB1010 also incorporates Rotel's own 'non-switching' output stage which, in common with most of the other pseudo-Class A arrangements, has a dual bias system to prevent either half of the output stage from ever being turned off.

TEST RESULTS

The 1010 could certainly deliver the power into all the test loads, a figure of 225W being recorded into 4ohms whilst the tone-burst test into 2ohms showed nearly 480W: so this amplifier would seen able to drive most loudspeaker loads. Further evidence of this was the extraordinary figure of 40amps being measured for peak current capability — well off our scale!

Harmonic and IM distortions were almost negligible — even at 100kHz the distortion was low at 0·3%. The absence of high-order high harmonics in the spectrum analysis confirmed the effectiveness of the non-switching bias circuits.

The disc overload capability was high and the stage was very quiet, a figure of 81dB being recorded for the weighted signal-to-noise ratio. The crosstalk, though, could have been improved, especially at 20kHz where only 34dB was measured. The equalisation, whilst fairly accurate, did show a slight treble boost, although this would in practice be partly counteracted by the rather high disc input capacitance which, at 390pF, would with many cartridges cause the frequency response to be rolled off at high frequencies.

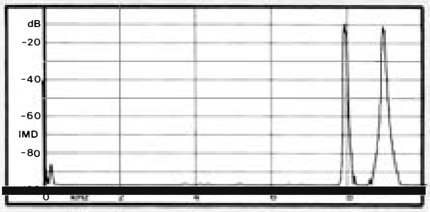
The overall sound quality was smooth with a slight midrange prominence. The sound stage was clear, although not too detailed. However, at high levels, this combination seemed to lack any power at low frequencies, and the bass became 'soggy' and without control. The sound was never unpleasant but high-level operation proved to be this amplifier's weak area.

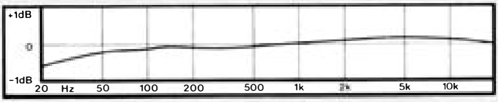
THE VERDICT

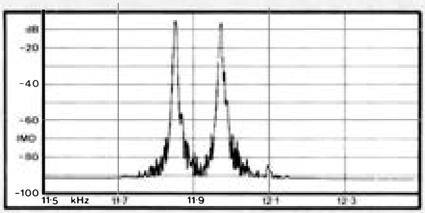
On paper a powerful amplifier and a combination offering good sound quality with excellent tape recording facilities. However, in practice the 1010 never sounded as powerful as it should, the sound sometimes showing a distinct lack of control. But, judged against its price, this Rotel system offers a great deal, and the RB1010 must be considered one of the cheapest sources of good quality watts. So, although there are some reservations, this Rotel pair does merit a recommendation.

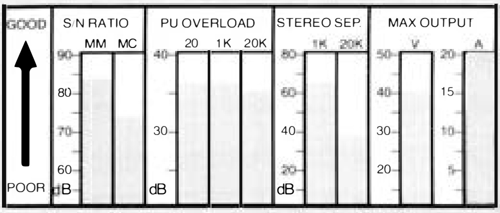
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	140	143	140
	8+8Ω	110	116	108
	4Ω	212	225	210
HARMONIC DISTORTION (% @ 100 WATTS)		0·006	0·004	0·005
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 6Hz		HIGH 67kHz









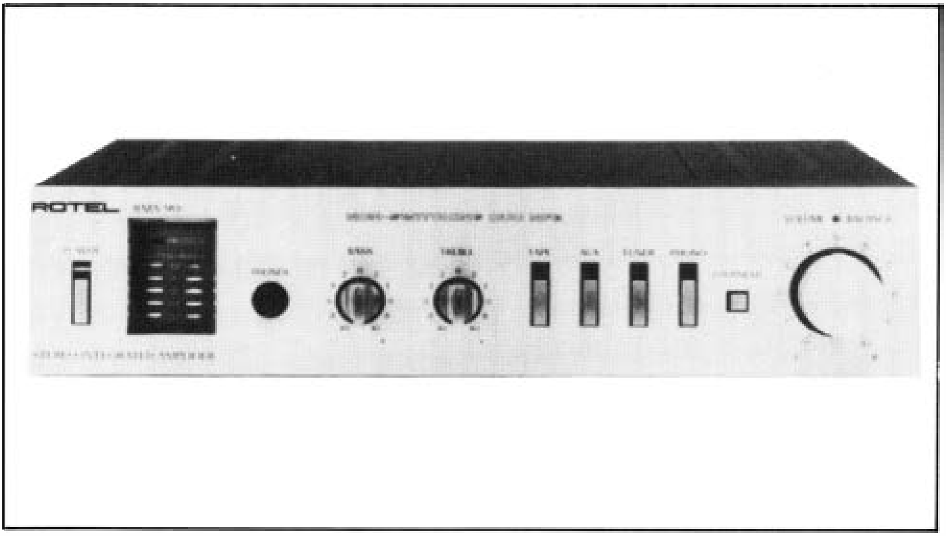
ROTEL RMA-90

The RMA-90 is the amplifier part of Rotel's latest micro system (the Micro 90) and as a result it has quite compact dimensions of about 280×44×260mm. Even so, it is a fairly powerful model, being rated at 40W per channel (8ohms). With such a small front panel, there is little room for lots of facilities, but it does have volume and balance operating concentrically, bass and treble controls (with detents at the zero position), loudness filter, and a headphone socket. There are three inputs and one for tape, these being selected by push buttons, each of which has an LED indicator.

Serving no useful purpose as far as I can see (but undoubtedly needed to match the competition) is a rather half-hearted power meter comprising five LED indicators per channel and telling you nothing that your ears couldn't do more cheaply. More useful is another indicator which glows red to tell you that the protection circuits, and hence the speaker relay, have operated. This is a help in those maddening periods of silence when you wonder whether to reach for fuse wire, soldering iron or telephone so that you can moan at the dealer who sold you a duff one.

Two pairs of loudspeakers may be connected and independently switched, but only through the use of two 'difficult to get at' slide switches located on the back panel.

The RMA-90's styling is fairly ordinary with an anodised aluminium front panel and dark grey paintwork on the rest of the case giving a smart but unobtrusive appearance.



CONSTRUCTION & TECHNOLOGY

It gets harder as the years pass to criticise the constructional quality of Japanese amplifiers, for production engineering is one of the areas in which they leave the rest of the world standing. The Rotel is no exception. Although the chassis is obviously crowded, it is laid out neatly. The chassis is split by a large extruded heatsink and I was pleased to see that Rotel had resisted the urge to make this too small. Had they done so, they would have saved money and saved valuable internal space, but the amplifier would have run hotter and perhaps less reliably.

The amplifier circuits mostly comprise discrete transistors with some effort having been put into the design of the power amplifier stages. The power amplifier design results in a fast, wide bandwidth circuit wth separate AC and DC feedback paths. To maintain good stability, the DC feedback is through a servo-amplifier to increase the gain and hence improve the sensitivity so that even a small DC offset voltage is corrected.

Finally, the RMA-90 has a non-switching output stage with two biasing networks to ensure that the output stage is never cut off, and hence avoid crossover and high frequency 'notch' distortion.

£95 MERIT ★★★☆☆ VALUE ★★★☆☆

TEST RESULTS

The RMA-90 just about met its power specification with 41W for both channel operations and 53W single channel (8ohm), whilst the short-term burst power (single channel) was a healthy 84W. The limiting factor was obviously the power supply which was of limited capacity in order to fit into the small box. So, although the power rose into 4ohms, it was only a moderate rise to 72W. However, the harmonic and IM distortions were very low and the amplifier was virtually unaffected by the capacitive load and gave almost its full output before the distortion rose.

The pre-amplifier stages returned good results with low noise figures and adequate stereo separation, but the input overload margin was rather poor at 20kHz. This could have been a natural consequence of using passive HF equalisation but this could not be confirmed. The bandwidth was rather too extended, being only 2·2dB down at 100kHz, there being no real justification for such a wide response.

The measurements of RIAA equalisation accuracy were initially poor due to a slight inaccuracy in the zero setting of the tone controls. When reset, the response stayed within ±0·35dB band from 20 to 20,000Hz and showed slight lower and upper midband emphasis.

The listening tests showed the RMA-90 to be an average performer. On the plus side the sound was felt to be clear and fairly open with good imaging, but the bass was less clearly defined and somewhat uncontrolled. Used with, for example, Rogers LS3/5s the overall sound was pleasant and fairly accurate, but with larger speakers the 'soft' bass was a distraction.

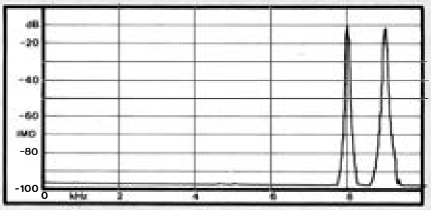
THE VERDICT

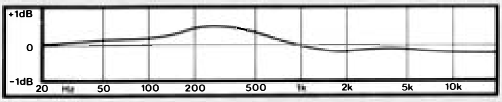
Well, at around £95 in the shops this small Rotel amplifier certainly offers fair value although you are obviously paying more for the complication of packing 40W into such a small package. Taken as a medium-power integrated amplifier the overall performance, both measured and subjective, is about average. But taken as a 'micro style' amplifier the RMA-90 is one of the best we've tried, its small size not resulting in undesirable compromises.

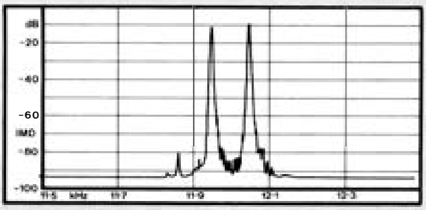
TEST RESULTS

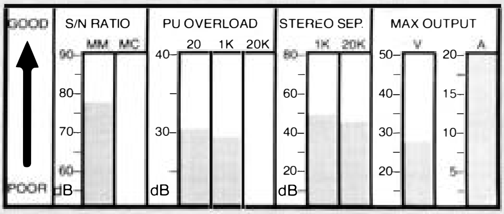
TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	52	53	53
	8+8Ω	38	41	41
	4Ω	70	72	72
HARMONIC DISTORTION (% @ 30 WATTS)		0·019	0·009	0·20

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	6Hz	74kHz









SAE 3000 3100

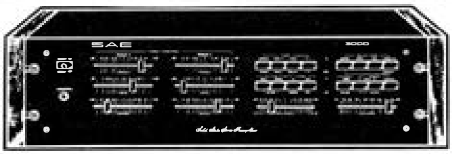
Audio manufacturer SAE was established some years ago in California. Unlike many American hi-fi companies, they take exporting seriously and have set up a European marketing subsidiary based in Italy. In their range they have five power and four preamplifier models with outputs rated from 50 to 400W per channel. The models submitted for review were their two most inexpensive models, the 3000 preamplifier and the 3100 power amplifier (50W). Together they cost about £460, a price which seems remarkably reasonable for a US import.

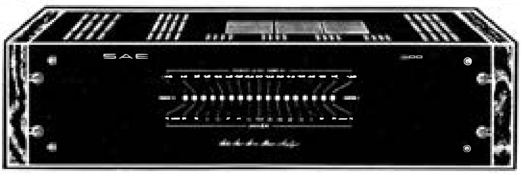
Both units are finished in a spartan black with white and blue screen printing. The styling is certainly not elegant but they do have a certain 'industrial technology' appeal. They look at their best in one of SAE's own racks but when used alone they can be domesticated a little by fitting the optional walnut sides.

The 3100 is a basic box lacking even a mains supply switch (in the US it would be switched by the preamp), but the front panel is occupied by a power meter array which uses eight large red LEDs for each channel to display the output signal voltage which is scaled as the equivalent power into 8ohms.

By contrast, the small 3000 preamplifier seems to consist mainly of a front panel absolutely crowded with controls. Four inputs (including two for disc) and two tape sources can be selected by push-buttons and tape-dubbing is possible in both directions. Complex mode switching offers not just mono or stereo but also Mono L or Mono R or stereo reverse — the latter useful for checking speaker phase. Separate slider tone-controls are provided for each channel, covering bass and treble (cut and boost) and mid frequencies (boost only centred at 1kHz). There are also two low filters operating at nominally 30 and 100Hz. Further switching ensures that either the normal signal or the tape signal paths can be through the tone-control section or direct in the bypass mode.

Finally, the preamplifier also has a socket suitable for medium to high impedance headphones.





CONSTRUCTION & TECHNOLOGY

To gain access to either unit involves removing an awful lot of self-tapping screws from the functionally basic casework. Internal construction is neat and straightforward and should keep life simple for the service engineer. The components are of good quality and of adequate rating, the output relay and coils in the power amplifier being well over-size for a domestic product.

The preamplifier is built up with three larger interconnecting boards so that the need for wiring is reduced to a minimum. Again, high quality glassfibre boards are used and a high standard of constructional quality achieved.

The power amplifier circuit is remarkably simple, being a fully complementary mirror-imaged design. Each half has a dual-transistor long-tail pair feeding a single transistor voltage amplifier which drives a Darlington output stage.

No circuits were provided, but it was noted that the 3000 uses ICs throughout and that the high frequency time-constant of the RIAA equalisation is provided by a passive filter at the output of the disc amplifier.

£229 MERIT ★★☆☆☆
£235 VALUE ★☆☆☆☆

TEST RESULTS

This combination was certainly able to deliver the power, with 81W continuous for a single channel into 8ohms and 105W for a short term burst. The power rose nicely into 4ohms, a figure of 131W being recorded. Similarly, the capacitive load was fully driven with ease so the SAE should be man enough for most loudspeaker loads.

Harmonic distortion was quite low at all audio frequencies and of low harmonic order.

The preamplifier section was found to be very noisy on all inputs even after applying CCIR weighting. Looking at SAE's own specifications, it was noted that the S/N of the model 3000 disc input is a claimed 81dBA ref. 5mV, but for their higher priced model 2100 a figure of 89dBA is quoted. Thus it would seem that some design compromises have become necessary to keep the cost down.

The pickup overload margin is fine but is slightly reduced at high frequencies. The stereo separation figures, whilst adequate, are a little low, particularly at 1kHz where a figure of 50 to 60dB would be more normal.

The frequency response is sensibly curtailed the −1dB points falling at 7Hz and 45Hz, whilst the RIAA response is accurate though showing a slight HF roll-off and LF boost.

The SAE also had an adequate peak output current of 17 amps with no pronounced protection effects.

The listening tests were initially unimpressive, the sound seeming to lack both clarity and an accurate reproduction of the dynamics of the music. Changing the preamplifier to the PS Audio model brought about a dramatic improvement, the power amplifier showing itself capable of delivering a firm detailed bass and an open midrange although the treble frequencies were rather too 'tizzy' for my tastes.

THE VERDICT

A rather unusual combination which seems to be aimed at the upmarket rack enthusiast and one which would definitely appeal to the wall-of-technology freak. The model 3100 is quite a good power amplifier offering good value for money and good sound quality.

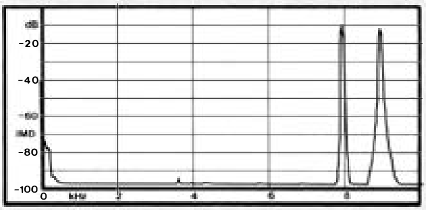
The model 3000 preamplifier is undoubtedly very flexible and offers a lot of facilities, but its sound quality could do with some improvement. It wasn't a case of the sound being unpleasant in any way, only the revelation of what was missing when a better alternative was tried.

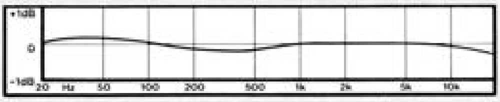
Overall, this SAE combination offers fair value and could be ideal for the frustrated recording engineers amongst us. But bear in mind its sonic limitations!

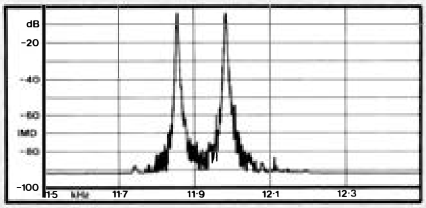
TEST RESULTS

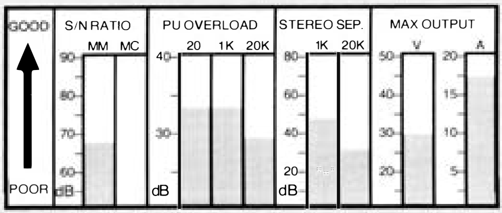
TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	76	81	81
(WATTS)	8+8Ω	64	81	81
	4Ω	121	131	129
HARMONIC DISTORTION (% @ 50 WATTS)		0·018	0·015	0·025

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	80Hz	45kHz









SANSUI AU-D9

The AU-D9 is the most powerful of Sansui's amplifier range, being rated at 95W per channel. As with the other models the AU-D9 incorporates the super-feedforward technology which uses error correction to improve the overall linearity; the designer's aim to achieve a virtually zero distortion amplifier.

This Sansui model is quite comprehensively equipped with inputs for both moving-coil and moving-magnet sources; and line inputs for tuner, aux and two tape decks with comprehensive switching to give cross-dubbing in both directions. The AU-D9 is equally well equipped with tone-controls and filters. Both the bass and the treble controls have switch selectable turnover frequencies, these being 3kHz and 6kHz for the treble control and 300Hz and 150Hz for the bass control. Two filters are provided — a 16Hz low-filter and a 20kHz high-filter. The tone controls can be bypassed by a tone cancel switch, but the filters must be switched individually. Two pairs of loudspeakers can be connected (and are individually switched) as well as a pair of headphones.

The front panel is a black finished aluminium extrusion with white and green lettering. It is dominated by the large diameter volume control which is not detented in the current fashion but does turn smoothly around a graduated scale. The steel case is covered by a wooden sleeve finished in an attractive pseudo-rosewood.

CONSTRUCTION & TECHNOLOGY

The AU-D9 is built to the high standards we have come to expect from the specialist Japanese manufacturers. Good quality components are used throughout and considerable care has been taken to fully screen all low-level paths and circuits. The two power transistors in each channel are mounted on a tubular heatpipe which transfers the heat away from the circuit boards. A single large transformer feeds a separate power supply for each channel, and a low-noise regulator to power the preamplifier circuits. The mains wiring was tidy and well insulated with additional shrouding much in evidence.

The circuit only uses discrete components in the signal path. A separate moving-coil head amp stage is fitted and this has an unusual mirror-imaged configuration. The following moving-magnet stage has the familiar op-amp configuration using low-noise FETs at the input and with an equalisation network in the negative feedback loop. This stage is also DC coupled from the input to the output (eg, no coupling capacitors) and this needs to be extremely stable to avoid any significant offset voltages. This stability is provided by a second feedback loop which has a DC servo amplifier to ensure a very high order of DC negative feedback without affecting the signal frequencies. The signal passes from the equalisation amplifier through the volume control after which it can be switched through the tone-control stage or directly to the input of the power amplifier. In fact when using the normal moving-magnet stage Sansui have a potentially 'straight line' amplifier the signal going through one preamplifier stage, the volume control and one power amplifier stage.

The power amplifier uses cascode stages to improve its high-frequency performance and linearity. It also has the 'Super Feedforward' element which consists of an extra low-power ultra-low distortion amplifier which senses the distortion originating in the output stage and then adds it (through the use of a small coupling transformer), out-of-phase, to the output signal thereby cancelling the original distortion. Any small traces of distortion not cancelled are then (in theory) reduced to insignificance by the overall negative feedback.

£360 MERIT ★★★★☆ VALUE ★★★☆☆

TEST RESULTS

With a midband power output of 144W reducing only slightly with both channels driven, the AU-D9 obviously has good power-delivery. A reasonable increase in power was recorded with the 4ohm load although the fast-acting protection circuit opened the speaker relay before the capacitive load could be fully driven. As expected, the harmonic distortion was extremely low at all audio frequencies and power levels.

Equally good results were achieved when the preamplifier section was tested with low noise levels and good overload capability by the pick up stage. One outstanding figure was the stereo separation which was 70dB at both 1kHz and 20kHz.

The disc equalisation proved to be extremely flat but showed a little roll-off at the frequency extremes.

Measurements of peak current suggested that a figure of 17amps would be representative, although we did find that the protection relay system operated after no more than a couple of full current peaks.

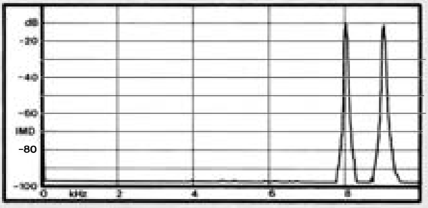
On audition the AU-D9 generally found favour. The tonal balance was about right with good overall integration. Furthermore the sound quality was particularly smooth and uncoloured with a fair amount of detail heard. The Sansui gave an open sound but without the rather artificial brightness that often accompanies this quality. This amplifier appeared to have two weaknesses, one being the moving-coil stage which gave the sound a rather constricted character and the other was the discovery that the AU-D9 changed when driving some loudspeakers known to be 'difficult' loads. However, with both the Quads and Studio Ones the overall sound quality was well above average.

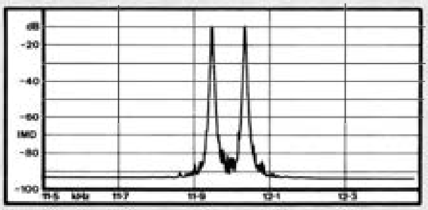
THE VERDICT

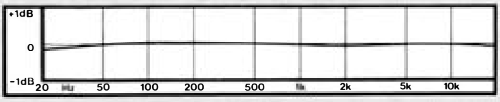
Without doubt this is one of the better 100W integrated amplifiers and one incorporating many interesting ideas. While achieving an above-average standard in both its measured and subjectively assessed performance, it is nonetheless quite expensive. But, hat said, it will be money well spent.

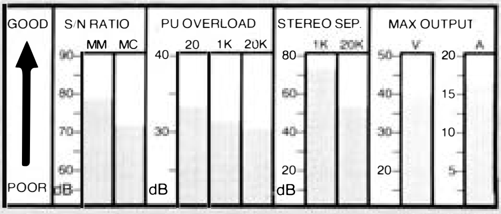
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	130	138	136
(WATTS)	8+8Ω	124	130	127
	4Ω	0·010	0·003	0·005
HARMONIC DISTORTION (% @ 100 WATTS)		0·010	0·003	0·005
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW	HIGH	
		DCHz	46kHz	







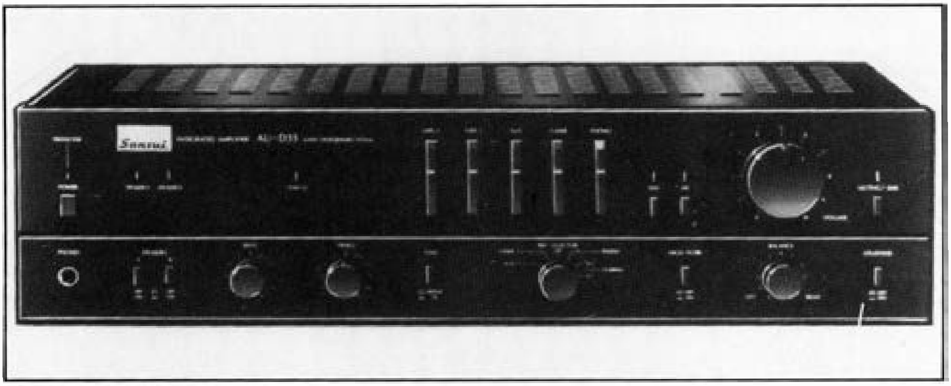


SANSUI AU-D33

When Sansui first introduced their 'Super Feedforward' technology I was pleased to test the first sample of the AU-D9 amplifier. I found its performance to have several pleasing features but being a 100W model it was moderately expensive. Sansui have now used the same type of circuit technique in their lower priced models, including the AU-D33 which is rated at 50W per channel and costs an equally modest £140. The finish achieved on this Sansui was a pleasant surprise. It is a genuine 'Made in Japan' model and it looks very nice indeed so you'll have less problems with the memsahib. The all-black finish is to a high standard and the front panel layout is balanced and well organised with tidy, clear white printing.

The AU-D33 is also quite comprehensively equipped having three inputs with separate selection of moving-coil or moving-magnet matching; and comprehensive switching provision for two tape recorders to allow cross-dubbing in either direction. A loudness circuit is provided together with a high frequency filter and the usual bass and treble controls which can be bypassed by a tone cancel switch. Connections are provided for two pairs of loudspeakers, which can be separately switched, and a socket is fitted for headphones. The volume control has none of the usual detented positions but does have a nice smooth action and is supplemented by a mute switch which reduces the signal level by 20dB. Operation of any of the five selector switches turns on an adjacent LED indicator to avoid any confusion.

All the input connectors are phono jack types whilst the speaker cable connectors are the push and twist types — convenient but difficult to use with the thicker loudspeaker cables such as QED 79.



CONSTRUCTION & TECHNOLOGY

Once the cover has been removed it becomes obvious that Sansui have not skimped on components or constructional quality in producing a budget priced amplifier. Indeed much of the construction is of a standard more commonly seen on higher priced models. For example a substantial heatsink, shielding around the pickup preamplifier board, plug and socket connection from boards to wiring looms for ease of servicing and the use of 'low distortion' electrolytic capacitors in the signal paths. The mains supply wiring was rather untidy but does meet BEAB standards.

At the time of writing no circuit diagram was available but a cursory examination revealed that the pickup stage uses a single low-noise integrated circuit whose gain and input loading resistor is changed by a cable-operated remote switch when the moving coil option is selected. The following line stage is a discrete operational amplifier using a low-noise dual transistor at the input. The power amplifier is non-inverting with a fully complementary output stage having the tone controls wired into the feedback loop. DC offset and short-circuit protection operates a loudspeaker switching relay which also mutes the output at switch-on. Operation of the protection circuit is shown by the front panel power indicator which flashes, and the supply must be switched off to reset the circuit.

The most important feature of the AU-D33 is its use of Sansui's so-called 'Super-Feedforward' principle. I have described this principle in the introduction but it may be helpful if I recall the

£139 MERIT ★★★☆☆ VALUE ★★★★☆

basics

Most conventional audio amplifiers use negative feedback to stabilise their gain and to reduce the distortion they generate in the signal. The feedback principle works by mixing the input signal with an out-of-phase portion of the output signal. The overall gain and the level of distortion is then reduced in proportion to the relative amplitude of this feedback signal.

Feedforward works by detecting just the distortion part of the signal at the output, amplifying it through a 'perfectly' linear small amplifier which is inverting and so feeds an inverted facsimile of the distortion back to the main amplifier's output. In doing so it causes a cancellation of the distortion. Sansui combine the two principles by applying feedforward error correction to the output stage of the amplifier then applying overall negative feedback to the whole amplifier to stabilise its gain and remove any remaining residual traces of distortion.

TEST RESULTS

Driving the 8ohm load the AU-D33 proved to be very generously rated single channel operation reaching 96W at mid-band frequencies — almost twice the manufacturer's rating. However the Sansui has a limited current capability (5 amps peak) as shown by its inability to fully drive a reactive load or the 4ohm load.

Harmonic and IM distortion products were both very low and the damping factor was very high, measuring over 200 right across the audio band.

The pickup equalisation proved to be exceptionally accurate for an amplifier of any price being within ±0·1dB limits from 40Hz to 20kHz.

Most of the other measured parameters gave average but not outstanding results. The high-filter was found to be a single pole type having too slow a slope to be much use (−3dB at 20kHz).

In common with most of the current Sansui range the sound through the moving coil input was rather disappointing, being metallic with a mid-range emphasis — in some ways the epitome of the 'transistor-sound'.

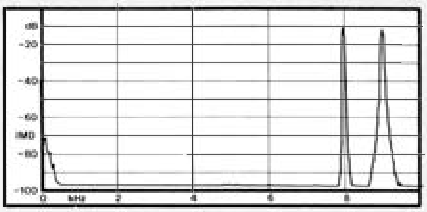
Far better results were achieved using the moving magnet input the sound being far better controlled; smooth (perhaps too smooth) in character with a firm extended bass. Some detail was lost but equally when used with indifferent disc sources there was some smoothing out of the rough edges, so in some installations the AU-D33 might prove to be a far better choice than a more informative design.

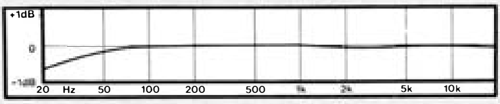
THE VERDICT

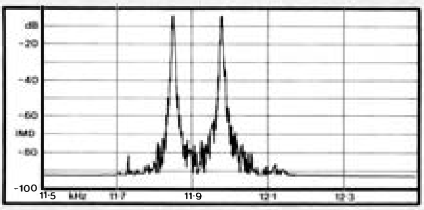
A well-engineered and nicely styled amplifier having much appeal. Although the loudspeaker choice needs a little care, this amplifier proved capable of producing a quite pleasant sound. At its present price the AU-D33 represents a good-looking, good value package and will undoubtedly be ideal for many hi-fi listeners.

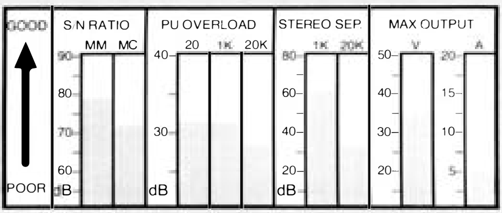
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	90	96	87
(WATTS)	8+8Ω	81	82	81
	4Ω	55	57	55
HARMONIC DISTORTION (% @ 50 WATTS)		0.003	0.002	0.004
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 6Hz		HIGH 32kHz







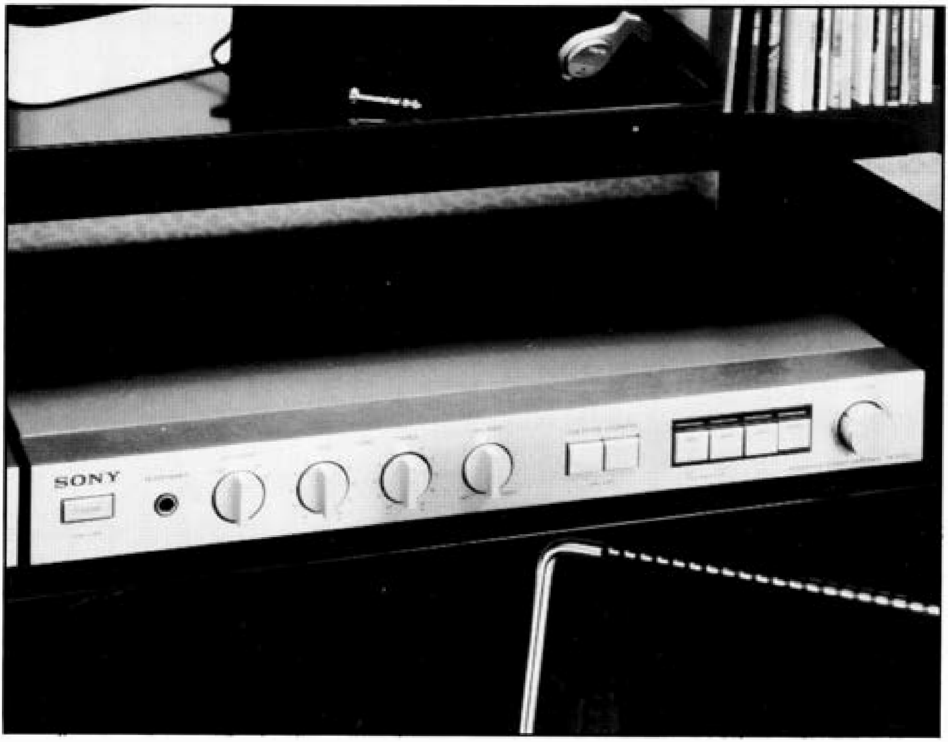


SONY TA-AX2

The TA-AX2 is Sony's smallest amplifier in both size and power. The slimline case is a mere two inches high with a neatly laid-out front panel. Whilst it is not unattractive the large 'chunky' push switches, although very practical, do give the TA-AX2 a low-cost appearance to match its price tag. Four inputs are provided, these being for moving-magnet disc, tuner and for two tape decks. Surprisingly, for a budget product, dubbing is possible in both directions between these decks. Two further switches operate the loudness and low filters (6dB/octave turning over at 50Hz). The bass and treble tone controls cannot be bypassed and they did prove to be slightly inaccurate in the 'O' position, the response not being completely flat.

A final rotary switch controls the output to the two pairs of loudspeaker outlets offering separate and both operation as well as speakers off for when the headphone socket is in use.

Its very simplicity made the TA-AX2 very straightforward to operate for, as I'm sure Sony would agree, there's a lot to be said for keeping hi-fi as humanly compatible as possible.



CONSTRUCTION & TECHNOLOGY

Internally the Sony is built to a high standard, although only the minimum number of components seems to have been used. However, despite the need to minimise manufacturing costs some care has been taken to improve potential reliability. For example, all cableforms are both soldered and glued to the printed circuit board to prevent any flexing and hence weakening of the soldered joints. Similar care has been taken to glue all bulky components down onto the boards.

The circuit design is very simple with a single integrated circuit being used as a disc amplifier, with a further single transistor gain stage in the tone control section. The entire power amplifier is one of the familiar low-cost IC power blocks. A protective fuse is fitted in the output line of each amplifier channel but it is an unusual PCB-mounted type and thus not readily replaceable by the average owner.

£90 **MERIT** ★★☆☆☆ **VALUE** ★★☆☆☆

TEST RESULTS

The power output showed a large difference between both and single channel operation and some drop-off at low-frequencies. The output into 4ohms showed some rise but only to 56W (1kHz). The limiting factor proved to be the regulation of the power supply which is necessarily small in order to fit into the slimline case.

The harmonic and IM distortions were both quite low in level and the frequency response was sensibly curtailed.

The noise level on the disc input was higher than average but this didn't prove to be obtrusive during the listening tests. The disc overload margin was also lower than usual but should be sufficient in practice. Finally the RIAA response. This proved to be only roughly accurate despite trimming the tone-controls for the flattest response.

All in all a fairly mundane set of results but at least no serious weaknesses were revealed.

Sadly, the first impressions given by the TA-X2 were not too encouraging. My feelings were how could the company who conceived the Esprit conceive this one! But then, the TA-AX2 is a considerably less inexpensive amplifier so I persevered on trying to be constructive.

The reproduction was flat with little location of depth and precious little in the way of accurate stereo imaging between the speakers. Indeed, the whole sound stage was in soft focus. When it was on the right, it was right; when on the left, it was left; and when it was halfway between, it was neither right nor left!

The Sony had only a limited ability to resolve detail and this was particularly evident when a vocalist had double tracked using ADT. Reproduced through this amplifier, it was difficult to isolate the two voices.

High frequencies had an audible colouration and a warmness that muddied the sound, the harmonics becoming indistinct.

THE VERDICT

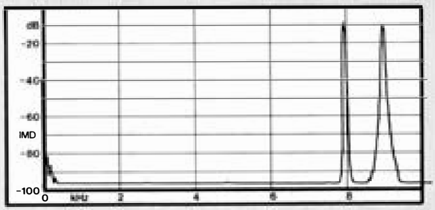
The TA-AX2 was pretty average. The measured performance brought it into the hi-fi league — but only just. Its sound quality was not particularly unpleasant — it just didn't reveal much of the music signals content or reproduce it accurately; but it was good enough to be hi-fi — but again only just.

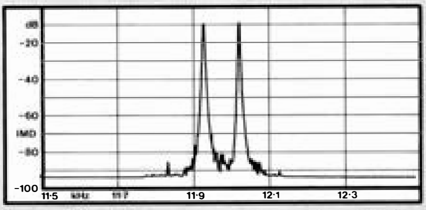
When compared to some of its contemporaries, it may be considered to be rather expensive for what it offers. If you like the long slim look without too many knobs and switches and can accept an inoffensive and undemanding sound quality, then this Sony amplifier could be just the one you're looking for.

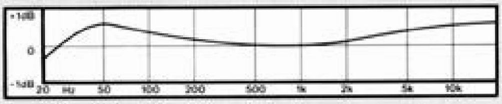
TEST RESULTS

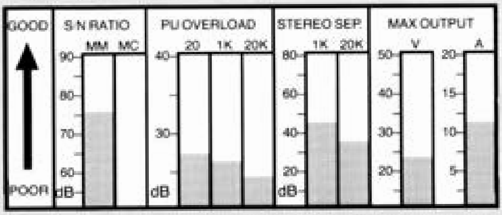
TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	41	44	43
(WATTS)	8+8Ω	30	36	36
	4Ω	51	56	54
HARMONIC DISTORTION (% @ 25 WATTS)		0·048	0·017	0·043

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	13Hz	35kHz









SONY TA-AX5

The TA-AX5 is an unusual amplifier, ostensibly a mid-priced 65W model, but having both a specification and front panel quite liberally loaded with features including Sony's Legato Linear and Audio Signal Processor (see later explanations). A glance at the highly technolgical front panel confirms that this model is well ahead in the high street features-for-price war but more importantly shows the need for a well-written owner's manual which fortunately Sony have provided.

The row of six flat metallic finish switches have a very light action and select the three inputs (including moving-magnet or moving-coil disc), the two tape inputs and tape dubbing from Deck 1 to Deck 2. The adjacent volume and balance controls are also touch pads pressure on either end changing the gain. The tone-controls are essentially bass and treble but each has a choice of three turnover frequencies and a pair of plus/minus touch pads in place of the more familiar rotary controls.

This amplifier also has a muting switch and two speaker switching but most of all it has the ASP (Audio Signal Processor) facility which permits two sets of tone control settings to be memorised and called back at will or replaced by a flat setting.

CONSTRUCTION & TECHNOLOGY

The interior of the TA-AX2 is very neat with most of the circuitry on a single PCB which is attached to a large double heatpipe cooler. Good quality components have been used; in particular some very nice push switches which have an excellent feel. Screened cable is used in all the signal paths although it is kept to a minimum through the use of remotely-operated switches. Nice touches are the small copper heat-clips to keep all the input transistors in good thermal contact with each other and the use of small RF toroids in the input lines to give some interference filtering — useful in these post CB days.

The power supply is of the switched-mode type so there isn't a conventional mains transformer, all the power supply circuitry being located in a small but heavily screened steel box.

The circuitry includes two novel features the first being Sony's 'Legato Linear' which in fact refers to a package of measures designed to make the power-amplifier very linear at high-frequencies with no loss of stability.

The second is the use of their ASP (Audio Signal Processor) in the preamplifier. This is basically an arithmetic processor and a non-volatile memory which register the volume, filter and tone control/settings and then adjust the amplifier performance by passing the signal through digitally controlled amplifier ICs. The claimed advantages are that only DC voltages and not signals are switched by the mechanical controls. Furthermore the memory enables two sets of tone-control settings to be pre-set and recalled at will.

£184 MERIT ★★★☆☆
VALUE ★★★☆☆

TEST RESULTS

This Sony had a healthy output into the 8ohm load, the single channel figure being about 100W compared to a rated figure of 65W. However the output into 4ohms was very disappointing falling to a mere 50W presumably because of current limiting by the protection circuits. Although the peak current capability was measured at 10 amps it would seem that significant non-linearities and, hence distortion, arose at a lower level.

The bandwidth was fairly well limited outside the audio band. The harmonic and IM distortions were more than adequately low, the small visible IM sidebands being of little significance.

All the remaining results were quite reasonable although the disc input headroom was rather low for amplifiers in this bracket and could be marginal with some high output cartridges.

When I first listened to this amplifier for an earlier magazine review I found little that made it impressive at its price. However, prolonged exposure has slightly changed my opinion (and incidentally highlighted another weakness of quick reviews). In many ways this Sony has a sound quality which is particularly relaxing without the soggy thickening of sound that normally characterises that adjective. Far from it — the bass was tight, if a little distant; the treble smooth and controlled; the imaging precise. Most of all the midrange had a sweetness of tone that was most endearing on some music. But the sound did lack a crispness on transients and the immediacy needed to fully convey the dynamics.

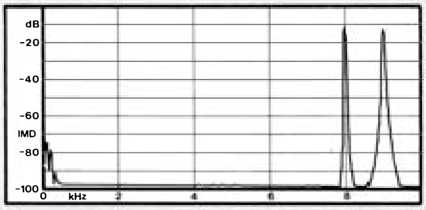
THE VERDICT

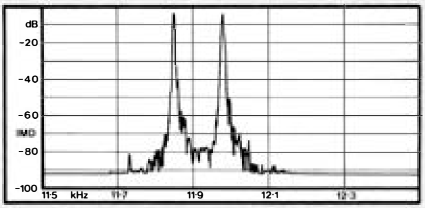
This is an interesting and unusual model and there is no doubt that some of the novel design ideas will become far more familiar in future years. However, as things stand the TA-AX5 is rather expensive for the performance offered. It will though undoubtedly have some appeal to those who revel in surrounding themselves with the fruits of our increasingly technological world.

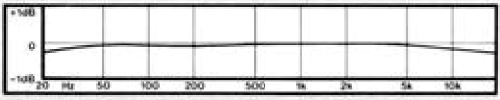
TEST RESULTS

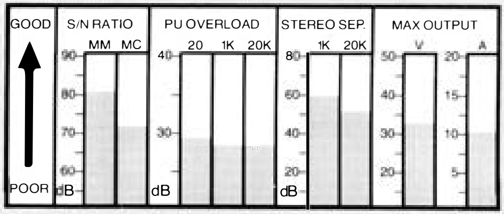
TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	98	101	98
(WATTS)	8+8Ω	89	92	90
	4Ω	34	52	52
HARMONIC DISTORTION (% @ 65 WATTS)		0·010	0·003	0·005

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	4Hz	50kHz









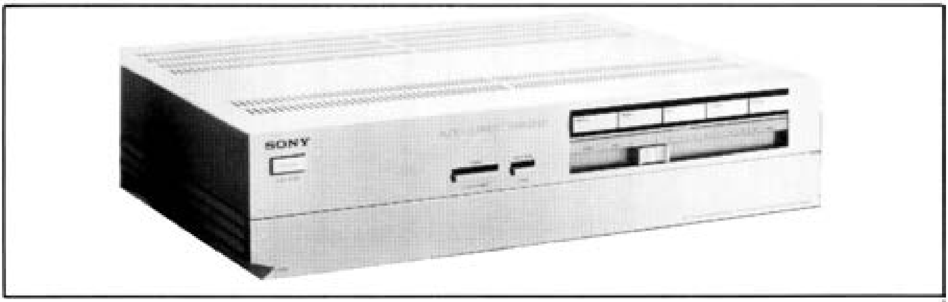
SONY TA-AX7

The TA-AX7 is one of the top models in Sony's TA range and is notable for its interesting styling and its incorporation of several new circuit techniques. This 80W per channel amplifier has a rather stark appearance, emphasised by its silver-grey finish. The front panel is dominated by the row of five input selector switches over a horizontal slider-fader style volume control. The rest of the panel is almost empty but in fact its lower half is hinged and smoothly tilts back to reveal an array of switches and knobs.

The Sony has three main inputs, and the pickup stage can accept both moving-coil and moving-magnet cartridges. There is a cartridge load selector which, for moving-magnets, will shunt the 50kohms load resistor with a 100pF of 330pF and, for moving coils, will select load resistors of either 100ohms or 30ohms.

Strangely enough, these two resistors ar selected by switch positions marked 40 and 3ohms, which presumably refers to the cartridge resistance. Bass and treble tone controls are provided together with a bass boost switch which gives about 7dB of lift at 20Hz and is recommended for those loudspeakers that are rather bass light. It should be remembered, though, that care will be needed to avoid rattling the voice coils of such speakers if the facility is used too enthusiastically. All these tone controls can be put out of circuit by using a tone cancel switch. Further switches are provided for muting (−20dB), low filter (15Hz turnover at 12dB/octave), mono-stereo, speakers (two pairs) and tape monitoring. The latter allows dubbing from one deck to the other in either direction.

There is also a headphone jack but, surprise, surprise, the lack of a loudness switch must make the TA-AX7 a rarity amongst Japanese products.



CONSTRUCTION & TECHNOLOGY

This particular Sony design incorporates a number of clever ideas intended to overcome what Sony believe to be the most significant weaknesses of conventional audio amplifiers.

The first technique is called Audio Current Drive. This is intended to overcome some of the problems associated with conventional voltage amplifiers, in particular the non-linearities caused by current flow through connectors and switches, and the crosstalk between channels which occurs because they share a ground path which has a finite impedance although in theory the ground line is a common point.

The theory of Sony's technique is quite complicated but in practical terms they convert the input signal voltage into a current which is then fed through the amplifier. As the signal voltage is now a signal current any error voltages no longer become part of the signal.

Secondly, there is the use of a linear gain control which has a law whose characteristic gives the best signal-to-noise ratio, best stereo separation and lowest distortion at low and middle positions of the 'volume' control. Sony say, quite reasonably, that most of us do not normally listen to music with the volume control cranked hard over, so why design the amplifier to work at its best in that position.

Next we have the quaintly named 'Legato Linear' which actually refers to a bucketful of ideas used in the power amplifier section. Fast output transistors are used to reduce the high-frequency switching distortion and the circuit arrangement conceived ensures low harmonic distortion.

Finally, before we pause for breath, Sony have incorporated a pulsed or switching type power supply which is small but offers good regulation. An unusual shunt regulator supply is used to power the preamplifier circuits because this represents almost a short-circuit to common-mode signals

£235 MERIT ★★★☆☆ VALUE ★★☆☆☆

and so prevents interaction between the two channels.

The TA-AX7, with covers off, reveals itself to be anything but a budget amplifer. A wealth of good quality parts is used to make up an unusually complex circuit arrangement. There is rather a lot of point-to-point wiring and some of this is rather untidy.

A heat pipe is used to cool the output transistors and this is positioned under extensive ventilation slots in the case.

TEST RESULTS

Although the Sony met its powerful specification when driving an 8ohm load the power output actually fell into 4ohms in sharp contrast to the more ideal doubling. This result suggests that the TA-AX7 should not be used with difficult loudspeaker loads. The protection circuits also came into play when the capacitive load was driven, a mere 9 volts of 20kHz signal being achieved before gross distortion occurred. With a purely resistive 8ohm load such an output voltage would be equivalent to only 10W!

The distortion was very low and the amplifier was found to have a wide power bandwidth. The preamplifier section had very accurate RIAA equalisation but the pickup input overload capability was quite average and only just sufficient if a high output cartridge is used. The signal-to-noise ratios were fine on all inputs except for moving-coil which was rather noisy.

However, the most extraordinary result was for stereo crosstalk where (at 1kHz) a staggering 84dB was recorded, while at 20kHz the figure was 62dB — normally considered a good result at 1kHz!

One operational problem did arise to cause some initial confusion. The tape monitor switches are not the usual press-on press-off types so, having used the monitor switch, it is necessary to re-press the source switch (eg phono) to listen to the original programme.

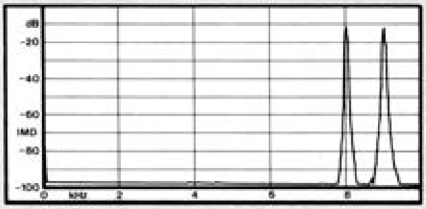
The TA-AX7 was not thought to be that impressive during the listening tests. The sound was quite smooth, even sweet, but some resolution of detail was felt to be absent. Nor was the Sony particularly good at reproducing the dynamics of the music. The most notable weakness was a certain lack of control, and hence tightness, at low-frequencies. With the Rogers Studio Ones the sound was reasonably satisfying.

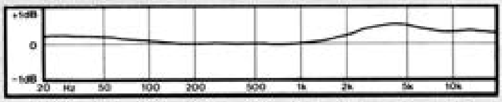
THE VERDICT

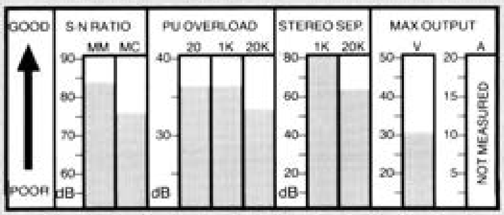
Despite the use of some quite interesting new technology, the performance of the TA-AX7 was about average for its price class. The claimed stereo separation was indeed remarkable, but the other measured results were quite unimpressive. The sound quality was quite pleasing with 'easy' loudspeaker loads although it is not an 'informative' amplifier. All in all, one of these '5 out of 10 for trying' products.

TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	97	104	100
	8+8Ω	92	95	94
	4Ω	83	87	83
HARMONIC DISTORTION (% @ 80 WATTS)		0·004	0·004	0·005
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 2Hz		HIGH 180kHz



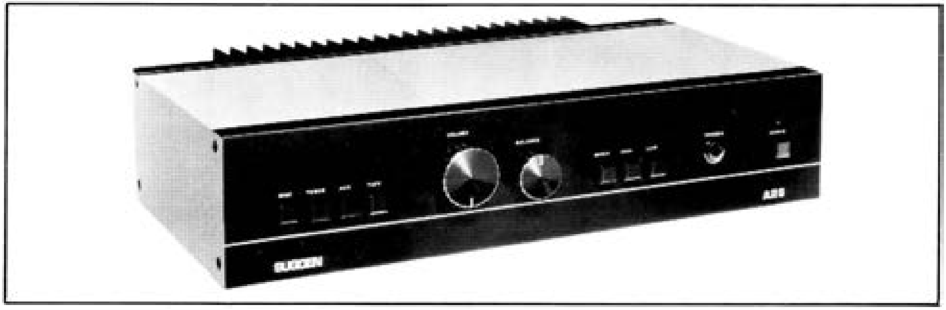




SUGDEN A28

The name Sugden seems to have been part of British hi-fi as long as most of us can remember, yet the company only produced their first amplifiers in 1967. It all started with Jim Sugden's first Class A design (the A21) which, if memory serves me right, had a power output of around 12W. Gradually, though, rather more conventional Class AB designs took over with models such as the A48 establishing a certain following. The latest model, the A28, takes them back to their roots for it is again a simple Class A design of lower power output — nominally 20W (8ohms). Now I just happen to be a firm believer in the merits of true Class A operation; I've designed several such amplifiers, and I could fill all the pages of this magazine with technical justifications for my beliefs. I was then a little disappointed to discover that the A28 is not a *true* Class A amplifier but yet another of the pseudo-Class A or 'non-switching' amplifiers. In Sugden's case they use the expression SQC which is short for 'Switched Quiescent Current' and this refers to an arrangement whereby the power amplifier runs with a standing current that ensures Class A operation up to 10W (8ohms), eg, about 9 volts RMS. Then when an incoming signal is detected that is going to cause the amplifier to exceed an output level equivalent to 7W, a transistor switch is operated which causes the standing current to be increased to 1·2 amps to ensure Class A operation up to 20W. Thereafter, the amplifier works in Class AB up to clipping at around 35W. Now I hate to be pedantic but either the A28 is a 10W 'pure' Class A amplifier or it's a 35W Class AB amplifier. But a switched bias amplifier is not a pure Class A amplifier.

The A28 is of straight line design, ie there are no tone controls so the front panel is rather short of knobs. Four inputs are provided (including one tape connection) and these are selected by a row of push buttons. The disc input is for moving-magnet cartridges but an optional moving-coil module will become available shortly. The volume and balance controls are followed by a mono switch and the high and low filters (both proper 18dB/octave types turning over at 10kHz and 30Hz) respectively. A headphone socket and power switch complete the front panel.



CONSTRUCTION & TECHNOLOGY

Virtually all the circuitry of A28 is built on a single large double-sided board which has a ground plane over its top surface. Some evidence of 'last-minute' modifications was found, but generally the construction was tidy with the use of good quality components throughout. Next to the input sockets the main board carries a 5-pin DIN socket into which was plugged a link, so presumably this is where the MC module goes. Each power amplifier has its own power supply and these are derived from separate windings of a common mains transformer.

The circuitry is essentially very simple and is much the same as that previously used in the A48 model. The disc amplifier is a single-ended design with RIAA equalisation using series negative feedback and selectable gain of either 28 or 59 times; a passive filter at the output extends the RIAA response beyond 20kHz.

The power amplifiers also work from a single positive rail and as a result they are AC coupled to the load through 10,000μF capacitors. The output stage is fully complementary and uses a pair of rather slow TIP plastic power transistors. Voltage gain is provided by a two stage AC coupled amplifier, the first stage of which is bootstrapped to the final output to improve linearity. The actual quiescent current switching circuit is common to

£195 MERIT ★★★☆☆ VALUE ★★★☆☆

both channels and consists of two op-amp stages, the first of which sums the two input signals to the left and right channel power amplifiers. The second stage is a comparator whose DC output changes polarity when its input exceeds a preset level. When it does so, it turns on a transistor connected in the bias chain of each channel and so increases the standing current.

TEST RESULTS

The A28 gave an average performance in the laboratory tests with no particular weak areas revealed, but equally no outstanding results. The power output into 8ohms at 1kHz was fine with 39W and 56W figures for both and single channel operation respectively. This further increased to 64W for one channel driving the 4ohm load. The harmonic distortion was moderately low although it rose significantly at high frequencies. The distortion was also virtually constant with output level over much of the operating range and was mostly third harmonic.

Turning to the preamplifier stages, the RIAA response was accurate up to about 12kHz after which a progressive roll-off began reaching −1dB (−0·8dB other channel) at 20kHz. The overload and noise figures for the pickup stages were adequate albeit unspectacular by present-day standards.

The IM distortion was rather poor with some quite noticeable sidebands revealed. In practical terms I would expect this non-linearity to result in a 'muddying' of the sound quality when listening to complex music and a lack of clarity in, say, choral works.

First impressions after a long listening session were of a solid sound quality to match the styling! The bass was well controlled and surprisingly well detailed, but the A28's best characteristics were shown in the midrange which was delivered with a smoothness which was most pleasant to hear. Higher frequency signals were clear but seemed a little distant, this tending to change the perspective on some recordings. Overall the effect was of a smooth but slightly 'soft' sound.

In the course of testing the A28 was found to have rather poor stereo separation figures, but quick work by the Sugden team resulted in a 10dB improvement overall.

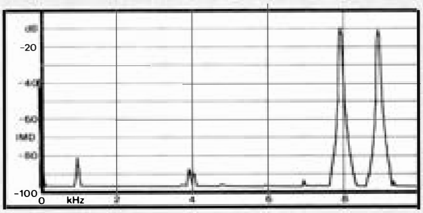
THE VERDICT

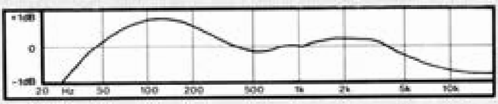
This new Sugden amplifier has some weaknesses, particularly its linearity at high frequencies, but it is still likely to prove popular. Generally the sound reproduction was smooth, relaxing and well integrated, although there was a loss of some detail and a lack of overall clarity. The A28 did tend to sound better when used with a slightly bright pair of loudspeakers.

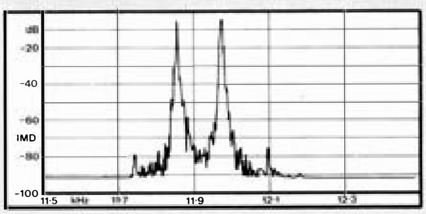
Against comparable Japanese product the A28 is rather more expensive but does offer itself as the genuine handbuilt British model.

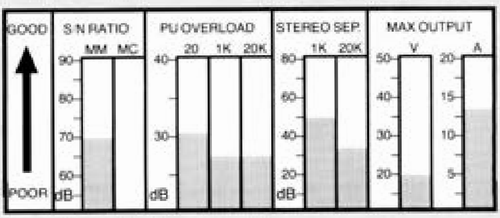
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	52	56	54
	8+8Ω	32	39	39
	4Ω	56	64	56
HARMONIC DISTORTION (% @ 25 WATTS)		0·042	0·014	0·050
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 16Hz		HIGH 24kHz









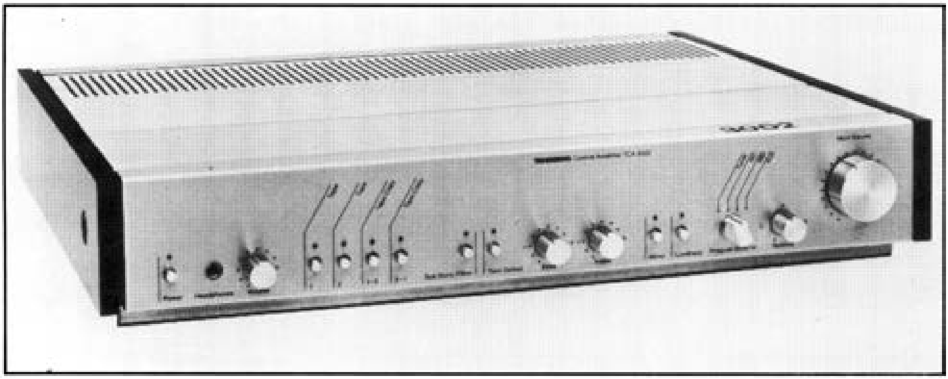
TANDBERG TCA3002 TPA3003

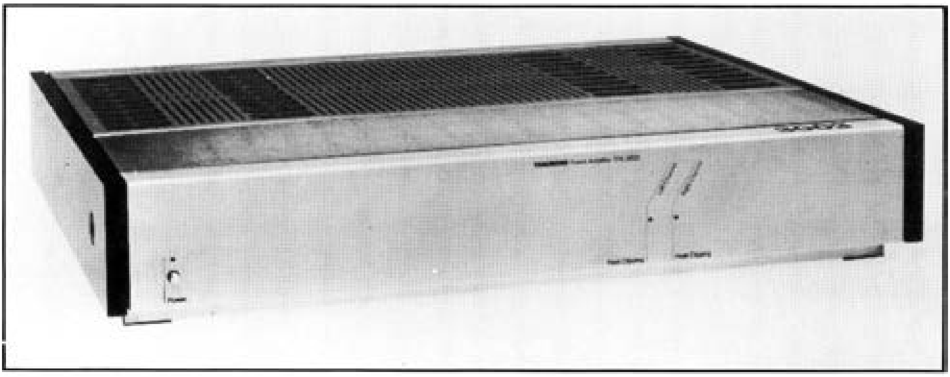
The Tandberg 3002/3003 is the pre and power amplifier combination from the new 3000 series which this Norwegian company has produced after its enforced reorganisation. The styling is restrained yet obviously Scandinavian, with some clever case design making use of interlocking aluminium extrusions. The front and rear panels are finished in a sort of metallic grey (with 'silver' control knobs and buttons) while the top surfaces are left as natural anodised aluminium. The 150W power amplifier has a plain fascia except for a mains power switch and two LED indicators to show peak clipping.

The preamplifier is comprehensively equipped with inputs for both moving-magnet and moving-coil cartridges, two tape decks with provision for dubbing and, unusually, a headphone output with its own volume control. The 3000 has bass and treble tone controls (and a tone-defeat switch) as well as loudness and subsonic filters — the latter particularly proving to be very effective is use.

All the rear connections are via phono sockets instead of the usual European DINs and, although a good DIN connector makes a better contact, most people do now prefer the convenience of phonos.

Alongside the disc inputs, the Tandberg has two small switches which change the moving-magnet input impedance (100/47/33kohms) and input shunt capacitance (20/120/350pF). The moving-coil input, though, has a fixed input loading of 1kohms which is unusually high but easily reduced by using adaptor plugs or by soldering parallel resistors acrosss the incoming cables.





£399 MERIT ★★★★★
£499 VALUE ★★★★☆

CONSTRUCTION

These two units are assembled to a very high standard. The cases are built substantially out of aluminium extrusions which lock together in an ingenious fashion. The 3003 layout is symmetrical with the power supply in the centre and one power amplifier module on each side. The amount of wiring has been reduced to a minimum by arranging for each PCB to plug into the next. The board layouts are very neat with all the component designations being screened onto the boards.

The preamplifier design uses completely separate amplifier stages for the MM and MC inputs although both follow roughly the same form. A linear buffer amplifier provides gain and then feeds through a passive HF filter to a second amplifier which has the remaining (bass boost) RIAA equalisation in its feedback loop.

These preamplifier stages are very fast and adopt an unusual form of compensation network, a series inductor in the transistor emitters. The power amplifier input stage also has this form of compensation but is otherwise quite conventional in design.

TEST RESULTS

The 3003 met its power output specification into an 8ohm load, although only by a small margin. The power rose when driving the 4ohm load to some 270W, the limiting factor being the regulation of the small power supply. The protection circuits gave no trouble and only came into operation when the 2ohm load was used.

Both IM and harmonic distortions were quite low with the midband THD measuring a mere 0·002%.

The pickup overload capability was excellent at 36dB, and the hum and noise levels were very low although the noise from the MC amplifier had a high level of low frequency 'rumblings'. The RIAA response was accurately followed and showed sensible roll-offs at either of the audio band.

For once the subsonic filter, all too often an afterthought in many designs, was found to work well, the useful 12dB/octave slope reducing the response by 4dB at 15Hz whilst having virtually no effect in the audio band.

The sound quality delivered by this combination was well above average. The sound was thought to be clear and crisp with good reproduction of loud passages rich in transients. The bass notes were delivered with convincing solidity and control. If the Tandberg has a weakness it is a slight lack of precision at high frequencies which seem to change their character with changing level. But this is a subtle problem which doesn't detract from an overall fine result.

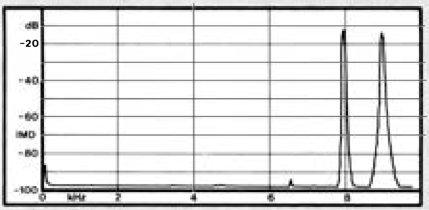
THE VERDICT

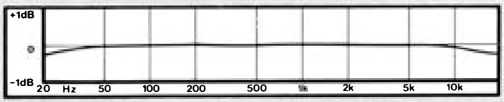
This Tandberg pair deserves high marks for a combination of design, construction and engineering excellence. Compared to their likely competitors, the Tandbergs represent good value for money and can be recommended.

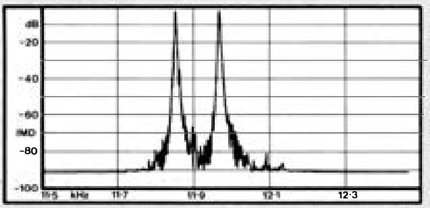
TEST RESULTS

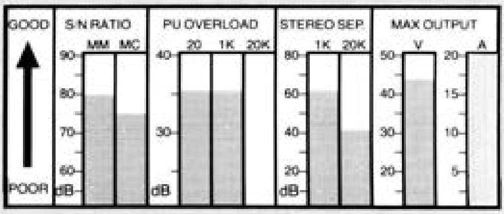
TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	200	206	200
(WATTS)	8+8Ω	178	184	180
	4Ω	270	280	272
HARMONIC DISTORTION (% @ 150 WATTS)		0·006	0·005	0·016

FREQUENCY RESPONSE BETWEEN –1dB POINTS	LOW	HIGH
	6Hz	80kHz





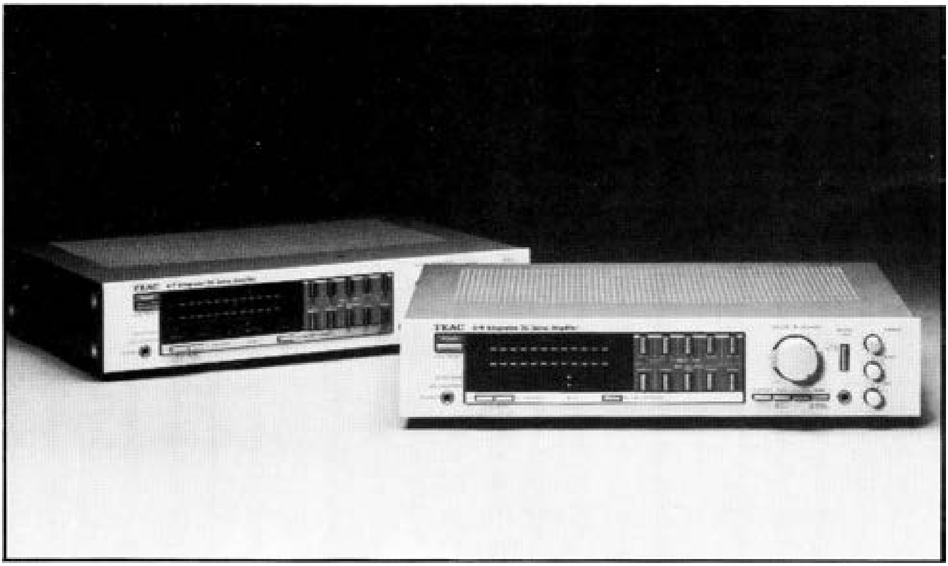




TEAC A7

Teac are not primarily known for their amplifiers having concentrated in recent years on their tape-recorders and phenomenally successful Tascam series of semiprofessional recording equipment. On the hi-fi side they have produced some interesting systems from one of which this A7 was drawn. The A7 is a 40W per channel model; the virtually identical A9 giving 60W. Although at first glance the control panel is a bit cluttered this amplifier is really quite smart and if you're into power-meters, LEDs and the like then the A7 does look the part. Surprisingly the front panel is very logically laid out. The top row of switches select the input source (three + two tape); the lower row select the recording source and allow tape dubbing in both directions. Switch selections cause indicators to light up in the segmented area under the power meter displays. The large knob is for volume (with concentric balance) the adjacent push button giving −20dB muting and the smaller knobs being bass, and treble. Further switches give a choice of moving-magnet or moving-coil cartridge matching, two speaker switching, loudness, stereo mode, subsonic filter and tone-cancel.

Finally, as might be expected from a company strong on recording, there is a microphone input socket and matching level control to mix with the selected source, although oddly enough it didn't seem to be possible to record from this mix.



CONSTRUCTION & TECHNOLOGY

The internal construction of the A7 is extremely untidy with lots of point-to-point wiring wandering under, over, and around the components. In fact the design shows few, if any, signs of the much vaunted prowess the Japanese claim in production engineering.

The common power-supply uses a small toroidal transformer secured in a steel screening can and a pair of rather small reservoir capacitors.

Generally the components are of a good standard but their assembly onto the PCBs could be done with more care.

All the preamplifier circuits use integrated circuits but the power-amplifier is discrete. Its circuit arrangement is simple with a dual-transistor at the differential input stage with an active load and a fully complementary output stage. The plastic power transistors are mounted on a generously rated heat sink.

The A7 also has DC and overload protection which operates the loudspeaker relay.

£139 **MERIT** ★★★☆☆ **VALUE** ★★☆☆☆

TEST RESULTS

For a nominally 40W amplifier the Teac delivered some good power output figures, nearly 60W into 8ohms (both channels driven) rising to 100W into 4ohms. The reactive load was also fully driven with no stability problems nor appreciable rise in distortion. In fact the distortions, both harmonic and IM, were very low. The damping factor was over 50 at all frequencies (over 100 below 2kHz) and the slew factor was more than 5.

The preamplifier set the bandwidth to sensible limits and also returned some good results with no problem areas. The RIAA response fell between fairly wide limits (+ 0, −0·3dB 50Hz to 20kHz) and showed a pronounced bass roll-off.

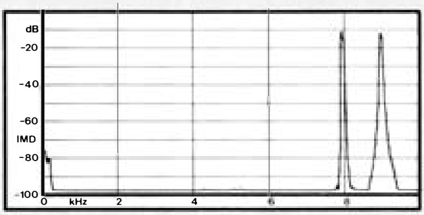
On audition the A7 proved to be quite a fair amplifier, the sound quality being controlled and forceful but perhaps a little ragged. The bass was firm and convincing but this amplifier had poor imaging which tended to put it into the wall of sound category.

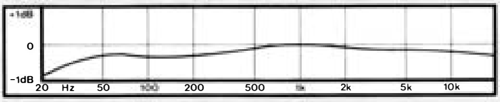
THE VERDICT

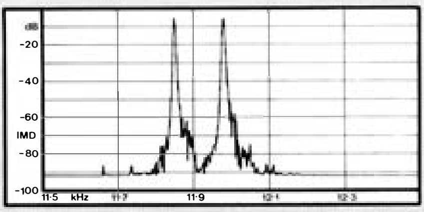
Although they are best known for their tape decks, Teac have proved that they are capable of making reasonable amplifiers. The A7 returned good results and produced a powerful but essentially 'middle-of-the-road' sound. While good on quantity and less so on quality the Teac A7 might be thought to be more expensive than its comparable competitors.

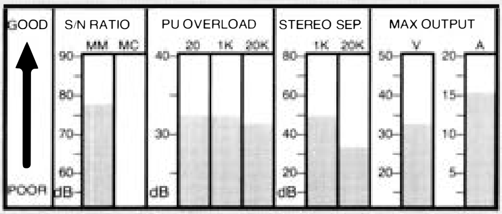
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	66	74	74
(WATTS)	8+8Ω	48	59	58
	4Ω	84	100	100
HARMONIC DISTORTION (% @ 40 WATTS)		0·007	0·005	0·029
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 15Hz		HIGH 48kHz









TECHNICS SU-V3

The SU-V3 is at the bottom of Technic's current range yet offers a power output of 40W per channel at a low price. The case is a slim, compact one, nicely finished and having a silver satin-finish front panel neatly laid out in a reasonably logical order. The front panel is dominated by the large fluorescent-type power meters and the handsome, large-diameter volume control. Three inputs (disc is moving-magnet only) and two tape inputs can be selected, and a second switch gives control over all the tape routing to permit dubbing. The SU-V3 is comprehensively equipped with tone controls, having bass and treble rotaries, and high filter, low filter and loudness push-buttons. A further button increases the sensitivity of the power meter but, in common with many recent designs, there is no Mono switching, although this can sometimes be achieved by shorting together the left and right channel tape-output lines.

Two pairs of loudspeakers are provided for; these can be individually switched and there is a headphone socket. In common with most of the current Technics range, the SU-V3 uses 'new Class A synchro-bias' which refers to another of this year's crop of 'non-switching' output stages whose biasing is arranged to prevent either half from ever switching-off and hence giving rise to crossover and other distortions. To emphasise the importance of this rather unimportant feature, Technics have placed a prominent badge on the front panel.

CONSTRUCTION & TECHNOLOGY

Most of the circuitry is loaded onto a single large PCB which also carries a large fabricated aluminium heat-sink. The single power-transformer is surrounded by substantial screening including a wrap-round steel cover. This, and the use of cable-operated switches, obviously shows the aim of reducing mains harmonics pickup to a minimum.

Assembly is neat and to a high standard with (as is invariably the case with Japanese audio equipment) good quality components in use.

The circuitry is simple with a mix of ICs and transistors in the preamplifier stages, the design being quite orthodox with no novel features. The power amplifier is also very simple with the economic use of discrete transistors with a complementary output-stage using plastic pack transistors.

£119 MERIT ★★★★☆
VALUE ★★★★☆

TEST RESULTS

The power output showed a significant drop between single both-channel operation, although the figures recorded showed a healthy margin over the rated 40W. There was an impressive near-doubling of power output into the 4ohm load but some current limiting occurred when a 2ohm was used. The output capability of the Technics was further verified by a very fine measured peak current of 23amps.

Harmonic distortion was low although there was a slight trace of sidebands in the two-tone IM distortion tests. The frequency response was rather too extended in the familiar Japanese mould, the high frequencies only being down 2dB at 100kHz.

The preamplifier stages generated quite low noise levels and had a good input overload capability (about 32dB) which held good across the band. The stereo separation, while good at high frequencies, could have been better at 1kHz. The disc input frequency response after RIAA equalisation showed a small degree of bass boost and treble cut, probably the result of a slight offset of the tone controls.

The Technics gave good results in the listening tests and, most important, seemed to be fairly tolerant of loudspeaker loading. The bass was rather forward and satisfyingly 'solid' in nature, although this control was soon lost at higher levels. At sensible listening levels, though, this amplifier produced a well controlled yet informative sound, free of the brashness characterising so many of the amplifiers in this price group.

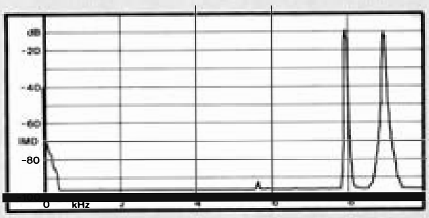
THE VERDICT

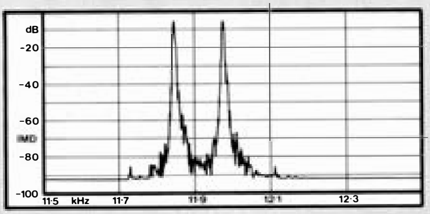
In many ways the Technics SU-V3 gives the lie to the widely held belief that Japanese amplifiers cannot produce a good sound quality. This amplifier was very pleasant to listen to and put many more expensive models to shame. It also gave a good account of itself in the laboratory with an impressive output capability being demonstrated.

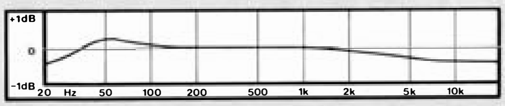
Amongst the budget priced models the SU-V3 offers a nice overall compromise between performance, facilities and appearance. It is to be hoped that Technics don't rush to replace this model for, as it stands, it is highly recommended.

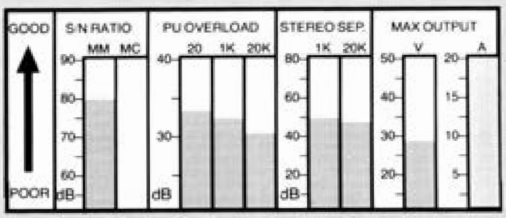
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	58	64	64
(WATTS)	8+8Ω	48	55	53
	4Ω	86	100	96
HARMONIC DISTORTION (% @ 40 WATTS)		0·011	0·013	0·04
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 8Hz		HIGH 85kHz







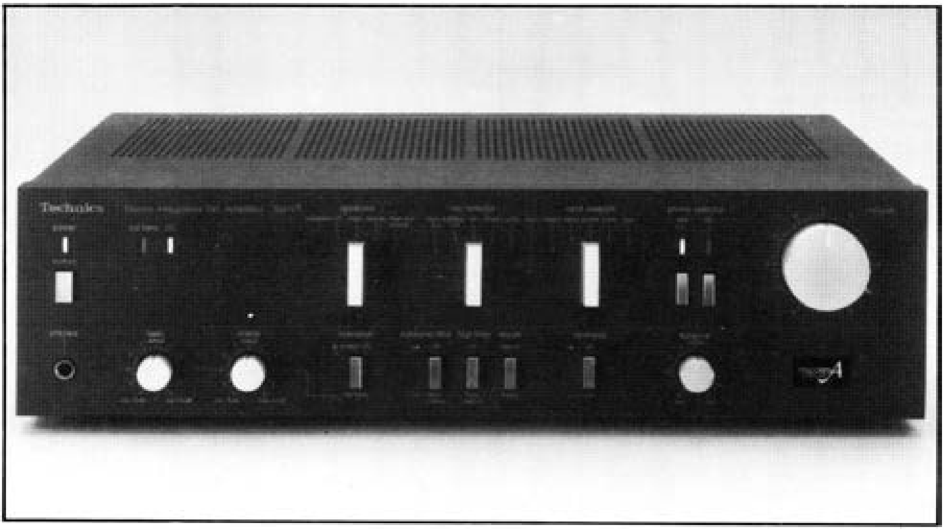


TECHNICS SU-V7

The Technics SU-V7 is a middle of the range model and is rated at 80W per channel. Below it comes the SU-V3 and SU-V5 rated at 40 and 60W respectively and above is the SU-V9, a heavy, expensive 120W model. The SU-V7, though, is quite competitive in price and is certainly attractive to the eye despite being fairly conventional in styling. The semi-black finish gives the product a more distinctive appearance than the regular 'brushed silver' seen on much of the Technics product range.

The front panel layout is uncluttered and businesslike being dominated by three selector switches (controlling input selection; tape switching and speaker switching) and a large diameter volume control which operates smoothly without any detents. The tape-switching for the two tape connections is versatile with recordings being possible from sources other than the one being listened to. So, for example, the radio signal can be recorded whilst a disc source is played through the speakers. By pressing a front panel switch the gain and matching of the disc input can be made suitable for either moving-magnet or moving-coil cartridges although no choice of input loading is offered.

Bass and treble tone-controls are complemented by high and low filters and the ubiquitous loudness; a tone-cancel switch can be used to bypass this section. All the switches and controls had a smooth yet solid feel which reinforced an image of discrete quality.



CONSTRUCTION & TECHNOLOGY

The cover lifts to reveal an extremely crowded chassis dominated by what seems to be an oversized transformer housed in a smart and effective diecast cover. The assembly is neat and tidy with well located screened cables being used for all signal paths. Particular attention has been paid to the safety of the mains wiring which is all well shrouded. Good quality components are used throughout, the volume control, for example, being a premium grade item

The circuitry of the SU-V7 is partly discrete transistor and partly integrated circuits. The disc amplifier uses low-noise FETs followed by an IC and further ICs give gain in the tone-control and filter stages. The power amplifer uses a dual FET differential input and cascode transistors. The complementary output stage uses a paralleled pair of output transistors and also has the dual bias system which Technics term 'New Class A synchrobias'. Like all non-switching output stages this one uses two bias chains; one fixed and one floating to ensure that the output stage is never turned off.

£199 MERIT ★★★★☆
VALUE ★★★★☆

TEST RESULTS

The rated power output was exceeded by a generous margin with little difference between single and both channel operation, this suggesting the provision of a large 'stiff' power supply. The power output almost doubled when the 4ohm load was connected but the 2ohm load couldn't be fully driven because the current limiting protection came into operation. Even so the SU-V7 proved capable of a healthy 17 amps peak current capability.

One curious result was the damping factor which was about 50 at 1kHz but fell to 17 at 20kHz. This is probably due to the use of a large output inductor and could result in a reduced output level at high-frequencies with some loudspeaker models.

The other measurements all gave fine results particularly those for stereo separation (52dB at 20kHz). The frequency response was, however, rather too extended at the top end and there could be some benefit in an earlier roll-off.

The RIAA response was quite accurate, dropping within a ±0·25dB band from 20Hz to 20kHz.

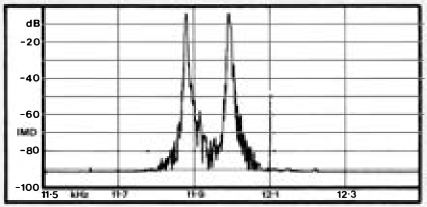
The SU-V7 did quite well in the listening tests, the sound quality being notably clear and detailed with excellent imaging. The sound was well controlled with good revelation of the dynamics of the music. The main weakness of this Technics model was at the bottom end which while accurate and moderately well detailed seemed to lack solidity at higher levels.

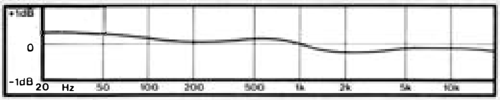
THE VERDICT

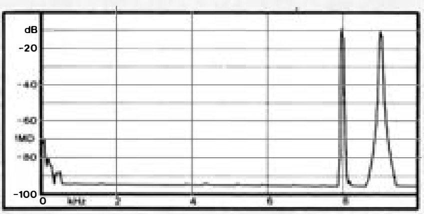
A very nice product and, for what is offered, not too expensive. The laboratory tests revealed little to criticise except perhaps the output impedance at high frequencies. Furthermore the listening tests again showed that some Japanese amplifiers achieve a sound quality which is highly satisfactory. The Technics SU-V7 is a fine amplifier and I do hope that it won't disappear in the annual product changes.

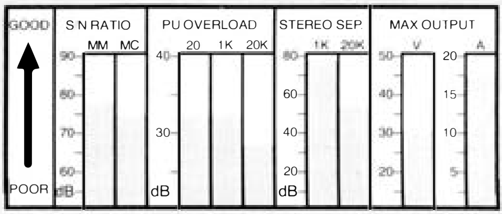
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	108	108	110
	8+8Ω	102	100	100
	4Ω	180	180	179
HARMONIC DISTORTION (% @ 80 WATTS)		0·08	0·07	0·05
FREQUENCY RESPONSE BETWEEN −1dB POINTS		LOW 6Hz		HIGH >150kHz





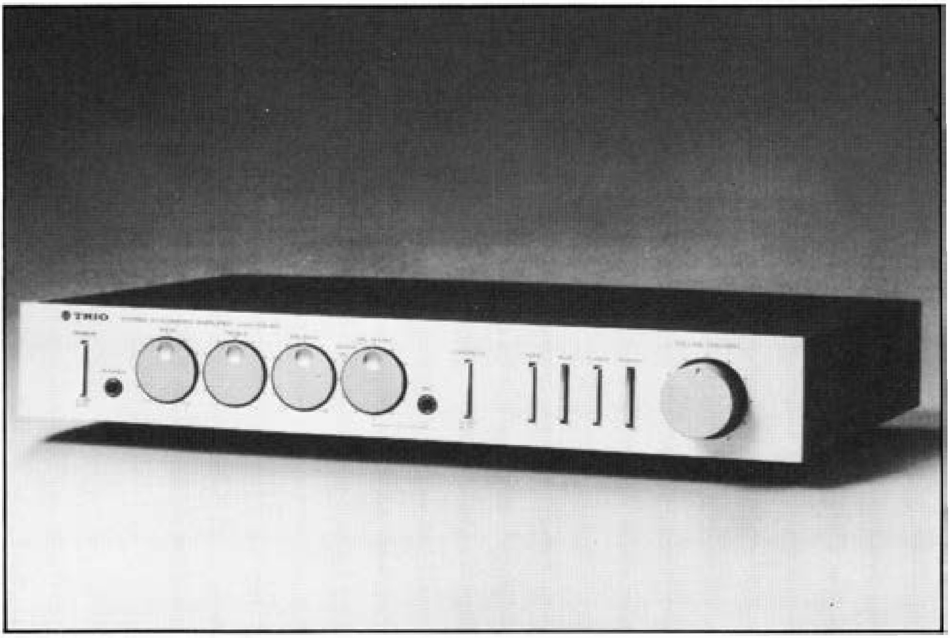




TRIO KA-60

The KA-60 is one of Trio's least expensive integrated amplifiers yet offers an interesting collection of facilities and a paper rating of 30W per channel. Finished in the usual silver grey the styling of this amplifier shows some attempts to be different with its use of flush control knobs. Although the controls are neatly laid out these knobs are an ergonomic disaster for they must be turned with a small finger-tip which does not permit smooth operation. Fortunately the volume control is conventional and a word of praise must be given to the selector switches which have a soft action and light-up when pressed. There is just the one tape connection but home recording enthusiasts may be interested in the microphone input (via a front panel jack) which can be mixed with the selected music source using one of the rotary controls. This control was found to mix the two signals smoothly cutting one off at each end of its rotation.

The only other facilities offered are a loudness switch and the bass and treble tone-controls which have a centre-indent as there is no bypass switch. Only one pair of loudspeakers can be connected and these are switched out of circuit whenever headphones are plugged in.



CONSTRUCTION & TECHNOLOGY

Internally the Trio is quite tidy apart from a few wandering wires which have not been adequately located. The chassis is split by a large piece of bent aluminium which serves as a heatsink and a 'power bulge' is necessary in the bottom plate to accomodate the transformer, which powers both channels.

The board assembly is fairly neat although there is a lot of surplus solder flux and residue which makes them look messy.

The circuitry seems very simple. A low-noise IC acts as disc input amplifier; thereafter all input signals are fed to the volume control and hence to the power amplifiers (a second IC is used as a microphone amplifier). The power amplifiers are equally simple: a long-tail pair input, a bootstrapped voltage-amplifier and a complementary output stage. In all just eight transistors and one IC per channel which must account for the low price.

£79 MERIT ★★★☆☆
VALUE ★★★★☆

TEST RESULTS

The Trio was capable of good power delivery for such an ostensibly small amplifier. The single channel power output was 56W into 8ohms rising to 77W into 4ohms and (for short periods only) 119W into 2ohms.

The distortion was acceptably low but showed an upward trend at higher frequencies resulting in a poor slew factor of 2½.

No major problems were revealed in the preamplifier section, good results being returned for overload margin, noise-figures, and stereo separation — the figure at 20kHz being very good. Also and surprising in a Japanese amplifier (and particularly 'High Speed' Trio) the overall bandwidth was quite tightly curtailed being 1dB down at 12Hz and 26kHz.

The RIAA response was reasonably smoothh and shows a gradual roll-off at the extremes.

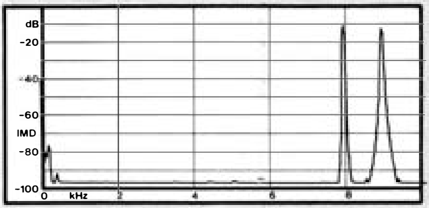
It might be tought that a £79 amplifier was unlikely to find favour in the listening tests but this Trio gained better marks than some models of twice the price. The sound quality seemed rather coloured and muddled in the upper bass/lower mid regions but there was good control, a sense of power, and reasonable smoothness at lower levels. It was more a case of detail missing than nasties added.

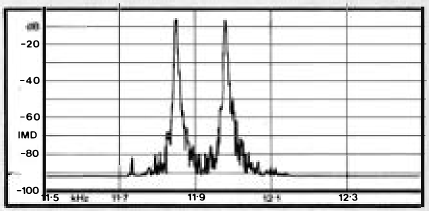
THE VERDICT

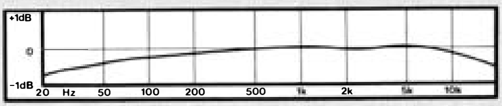
We've grown used to finding a lot of rubbish at the bottom end of the market but on this occasion Trio have done well. The KA-60 is a competent design having attractive styling and their customary high standard of finish. It returned a good measured performance and the sound quality was good enough to live with. In all very good value.

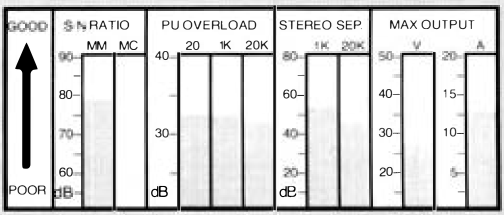
TEST RESULTS

TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	56	56	56
	8+8Ω	43	46	44
	4Ω	68	77	76
HARMONIC DISTORTION (% @ 30 WATTS)		0·031	0·016	0·047
FREQUENCY RESPONSE BETWEEN – 1dB POINTS		LOW 12Hz		HIGH 26kHz









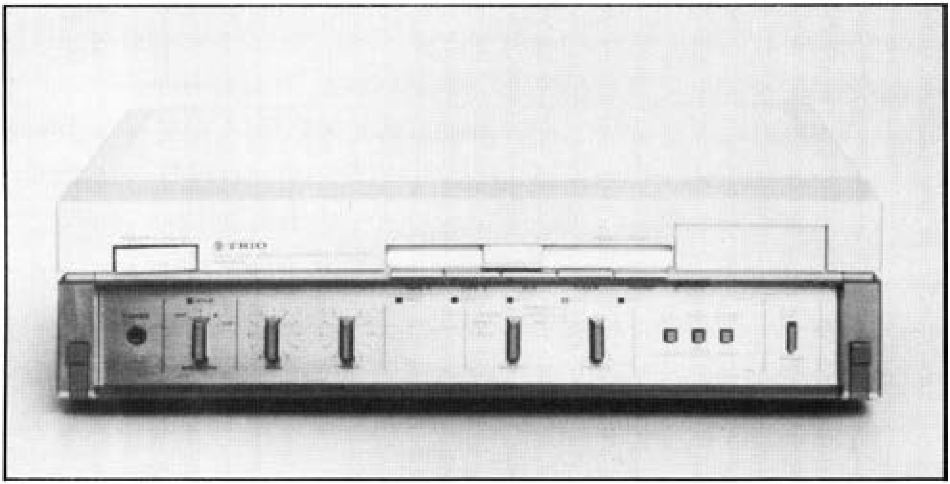
TRIO KA-800

This Trio amplifier is definitely one with which to impress the neighbours. It is housed in a beautifully turned-out grey plastic (non-magnetic!) case with the bottom half of the front fascia covered by a hinged clear acrylic cover which, in the best Japanese tradition, fits accurately and works smoothly.

With the lid closed the user has access to the three normal and two tape input selectors, the power switch, the horizontal slider level control and a big flat slab called FADER. You press this and over about five seconds the volume fades up to that set on the level control and then it glows greeny-blue. Press it again and the light goes off and the sound fades down. A lovely toy but believe me you're lost until you discover it.

Under the panel there are bass and treble controls, subsonic and loudness filters, balance, moving-magnet/moving-coil disc selection, record out giving tape dubbing in both directions, headphones socket and the speaker switching (two pairs) which also allows the Sigma Drive (see later) system to be turned on.

This 50W per channel amplifier is very nicely styled with a standard of finish that was a joy to see. What's more the blue fader continues to amaze my neighbours!



CONSTRUCTION & TECHNOLOGY

The KA-800 has a single, rather small, power-supply feeding both channels. The heatsink is a massive heatpipe to which the four plastic power transistors are bolted. Most of the circuitry is on one very large PCB which is of high quality and, most unusually, etched on a 'maximum copper' basis. There is a fair amount of wiring over the board and this could have been tidier. The mains wiring is unshrouded and, surprisingly, only the live line is switched so care should be taken to wire up the mains plug correctly.

The circuitry uses a mixture of ICs and discrete transistors, the disc amplifier, for example, having two low-noise FETs before an IC amplifier stage. The real novelty of the KA-800 lies in its use of 'Sigma Drive' in the power amplifiers. Put at its simplest this is an arrangement intended to compensate for the signal loss and degredation caused by the speaker cables and connections. So the sensing point for the negative feedback is moved from the output transistors to the loud-speaker itself. Four wires are used; two carrying the signal to the speaker and two others, also attached to the speaker terminals, which act as return or sense paths for the feedback. Then in accordance with the laws of feedback any signal loss or distortion occurring between the amplifer output and the speaker terminals will be reduced in proportion to the amount of feedback applied.

£225 MERIT ★★★☆☆
VALUE ★★☆☆☆

TEST RESULTS

The KA-800 had a generous power output into 8ohms, some 83W being measured for one channel operation but the power into 4ohms actually fell to 77W and the reactive load could hardly be driven at all without the protection circuits tripping out. The reason was not hard to find, for the KA-800 has a peak current capability of just 6amps but current of 2½amps drawn for under half a second causes the protection to trip.

The harmonic and distortions were low although some sidebands were visible and the overall bandwidth (Aux. to Spk.) was sensibly curtailed with −1dB limits of 8Hz and 26kHz.

The performance of the disc input stage was quite good but the stereo separation, whilst very good at 20kHz, could usefully be improved at 1kHz.

The RIAA equalised response was accurate between 50Hz and 20kHz and showed a small degree of tilt with treble boost and bass-cut.

First impressions of the sound quality were rather good, the clarity and imaging being paticularly creditable. At low levels the control was good, the bass nice and solid whilst the Sigma drive when switched in seemed to tidy up the sound stage to give such precise locations as the original recording techniques would permit.

However the KA-800 was prone to a worsening of sound quality as the level rose with a tonal shift to treble emphasis, possibly made more noticeable by the sogginess which afflicted the previously firm bass.

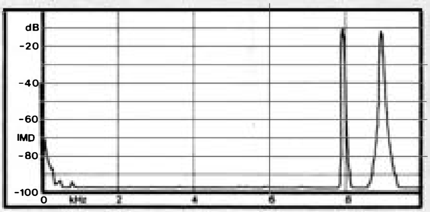
THE VERDICT

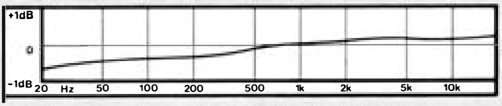
The KA-800 is an expensive amplifier but it is an amplifier that is a little different both in styling and operation. It works reasonably well and although the listening tests revealed one or two weak areas it did prove to be a popular amplifier when wired into the home hi-fi. In the right system it could well give you both listening pleasure and a conversation piece.

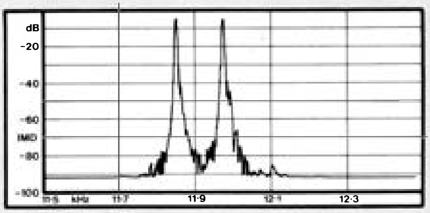
TEST RESULTS

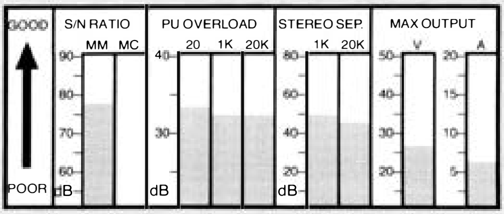
TEST		20Hz	1kHz	20kHz
POWER OUTPUT	8Ω	80	83	83
(WATTS)	8+8Ω	62	70	73
	4Ω	77	77	75
HARMONIC DISTORTION (% @ 50 WATTS)		0·011	0·004	0·015

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	8Hz	26kHz









YAMAHA A760

The A-760 is an 80W integrated design whose styling and appearance makes it immediately recognisable as Yamaha. A good range of facilities is provided including a moving-coil pickup input. Much effort has gone into a bar-scale power indicator where a slider control moves along a graduated 'power' scale until the LED mounted in the control knob starts to flash — very refined and very neat, but not all that useful. An equally sophisticated loudness control is provided in the form of a second volume control. The maximum desired listening level is set by the volume control; then, to reduce the volume, the loudness control is rotated. As well as a drop in overall level there is the compensating boost of high and low frequencies. Another novel feature is a large illuminated button labelled 'DISC' which, when pressed, overrides all the input switches and routes through the pickup input. A nice bit of user-convenience and possibly a more direct signal path as well.

The A-760 is finished to a very high standard, the overall appearance only being spoilt by the cluttered mixture of different sizes and shapes of control knobs.

CONSTRUCTION & TECHNOLOGY

Yamaha have managed to cram an awful lot of parts into the case. Good quality parts are used throughout with generous sized extruded heat-sinks and a large transformer which feeds both power amplifiers, although each has a separate rectifier and capacitor circuit to give semi-separate supplies. All the low-level signal switching takes place in the input socket area using cable-operated switches operated from the front panel. The A-760 is well built but does have an untidy internal appearance. The crowded nature of the chassis and the use of a fair amount of wiring is reminiscent of a rat's nest — but a better class of rat's nest!

Nearly all the preamplifier circuitry is based around the use of low-noise integrated circuits which tend to give a predictable level of performance at a reasonable manufacturing cost. The power amplifier is quite conventional and uses bipolar output transistors which are comprehensively protected by both current limiters and a relay system which disconnects the load in the event of an overload.

TEST RESULTS

In the laboratory the Yamaha A-760 was revealed to be a well thought-out design. The manufacturer's rated power-output was comfortably exceeded when driving an 8ohm load and the large increase in output driving the 4ohm load confirmed both absence of power limiting under these conditions and the quality of the power supply. Distortion, both intermodulation and total harmonic, was low and showed good linearity across the audio band. The preamplifier stages returned good signal-to-noise figures and the pick-up stage headroom was fine although it fell at 20kHz but this is of no practical consequence. The RIAA equalisation was very accurate showing a slight HF roll-off — but very slight, a mere 0·2dB. The stereo separation fell at 20kHz but the figure of 48dB is quite adequate for disc sources. On the test bench, then, a good all-round performance.

Unfortunately, the A-760 had to be returned to the manufacturer before the peak current test could be performed so no figure has been quoted. However, our earlier tone burst tests into a 2ohm load (310W at 1kHz for two cycles) suggested a figure of 14 to 18 amps would probably be of the right order.

Unfortunately this didn't hold true for the listening tests which showed the A-760 to have a slightly below average performance. Initial impressions were of a very full sound with plenty of weight and power, particularly in the lower registers. But the progressive hardening of the sound if the volume was raised brought out its inherent 'transistor amplifier' quality. The detail, particularly in the upper-mid region, was filled in to create the familiar 'wall of sound'. Used at low listening levels the sound quality was liked but at medium to high levels the sound quality became degraded in the ways described above. Played loud the A-760 coarsened the sound and all control was lost at low frequencies, the bass becoming very boomy.

£249 **MERIT** ★★★☆☆
VALUE ★★☆☆☆

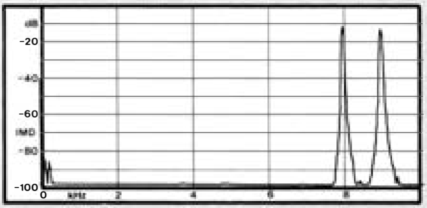
THE VERDICT

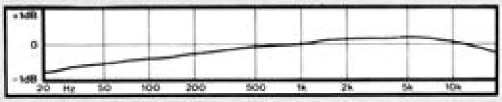
A difficult product to assess. The A-760 is a well-built amplifier with a high standard of finish offering a good range of facilities for the price. The laboratory results suggested a sound enough design, the figures being uniformly above average in this price group. But the listening tests left a big question mark. At low levels the sound was quite pleasant and drew little consistent criticism. But at higher levels the sound quality became poor — not particularly unpleasant but unsatisfying. This Yamaha may well suit many people but is not one for the 'play-it-loud' brigade.

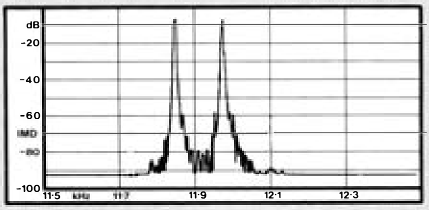
TEST RESULTS

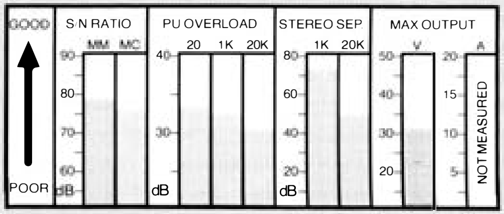
TEST		20Hz	1kHz	20kHz
POWER OUTPUT (WATTS)	8Ω	100	105	101
	8+8Ω	90	92	90
	4Ω	149	155	150
HARMONIC DISTORTION (% @ 80 WATTS)		0·006	0·005	0·008

FREQUENCY RESPONSE BETWEEN −1dB POINTS	LOW	HIGH
	5Hz	76kHz









MOVING-COIL STEP-UP DEVICES

The term 'Step-Up Devices' covers both transformers and the pre-preamplifiers more commonly known as head amplifiers. Although the output from some moving-coil cartridges can be fed directly to the moving-magnet input of the amplifier this is rarely successful for two reasons. First the noise level which may be 80dB (CCIR/ARM) below a 5mV signal but for a moving-coil cartridge the signal level drops to 0·5mV and so the noise is then only 60dB below the signal and probably quite annoying in practice. Secondly the input impedance of the normal (MM) disc input is of the order of 47kohms — a figure established in days of old to tame the mechanical resonances inherent in the early cartridges. However, moving-coil cartridges work best when loaded with lower impedances anywhere between 5 and 1000ohms, depending upon the manufacturer.

So the step-up device must fulfil three main needs. First it should provide signal gain: a figure of × 30 (30dB) being quite typical. Then it must present a suitable load to both the cartridge and the following preamplifier. Finally the residual levels of noise and distortion components should not, ideally, degrade the performance achieved by the moving-magnet preamplifier alone.

As to the frequency response of the step-up device. well there is much argument here, with some companies favouring extended responses. But my own work has indicated that this can lead to problems with the associated preamplifier (see later notes).

TRANSFORMERS

The transformer was the original step-up device, the earliest examples being little more than modified microphone transformers. At their simplest they have the advantages of low-cost, ease of operation and use, and freedom from any power supply. However, the performance of some of the lower-cost models was poor and there was a swing towards the use of active head-amplifiers. In the past few years, though, a number of exotic transformers have been produced using specialised transformer cores, complex windings patterns and often expensive wire using silver instead of high-purity copper.

The ideal transformer is thought by some designers to be the perfect step-up device. It is efficient, yet requires no power source; it contributes no noise other than the negligible thermal noise of the winding resistance; it adds no distortion; and being a passive device should be stable in operation and reliable.

The problems occur because there is no physical realisation of the ideal transformer and so some compromises set in, these, as always, being mainly a function of cost. First the transformer must have a wide bandwidth and because a passive device may only have a single-order roll-off, this should extend beyond the audio band at both extremes. A good LF response requires a high inductance which means a good (read, expensive) core and plenty of wire. An extended HF response requires a low interwinding capacitance so all those turns necessary for the LF response need to be wound in sections; a process which is inconvenient and therefore again expensive.

Distortion arises in two forms. First as the signal rises in level the core can become saturated, though this is rarely a practical problem with the very small signals involved. Secondly, and potentially far more seriously, the core is very non-linear at very low signal levels for it needs a small but finite current flow to create enough magnetic flux for the arrangement to start working as a transformer. Again careful choice of the core material minimises the problem.

Really there is no difficulty in building a good transformer on a 'cost-no-object' basis; the real skill comes, as is so often the case, in designing such a transformer at an affordable price.

Another legendary problem with transformers has been their susceptibility to hum induced by external fields from, say, power transformers. To a large extent the transformer can be shielded by a case of mu-metal or similar material but by far the cheapest form of shielding is distance and so many a hum-prone transformer has been silenced by placing it in a shoe-box.

Finally the transformer is rather inflexible in terms of input and output impedance. If it has a turns ratio of say 20 (eg. 100 primary turns and 2000 secondary turns) then the voltage gain will also be 20. The load impedance seen by the cartridge will be:

$$\frac{50,000}{20 \times 20}$$

(where the MM preamp input impedance is 50,000ohms) = 125ohms.

Thus the load impedance is directly related to the step-up ratio (or voltage gain).

MC HEADAMPLIFIERS

The head-amp is simply an amplifier stage providing the required gain and having input matching suitable for the MC cartridge. The performance of this stage in terms of distortion, overload margins, etc. need to be as good as found in normal moving-magnet disc amplifiers but the noise figure must be very low for the reasons described earlier. However, this can be quite a difficult task and various circuit techniques have been tried to minimise the noise. These all seek to minimise the total resistance (which generates noise) in the signal path and this includes the internal base resistance of the transistors.

One technique is to parallel several input transistors; another is to use small power-transistors which have an inherently low base spreading resistance. As well as the level of the residual noise it is important to pay attention to the noise spectra. Is the noise level essentially independent of frequency or does it increase at low-frequencies? The Fig. 2a shows the noise spectra of two amplifiers, one having a notably lower noise level than the other. But Fig. 2b shows the situation after RIAA equalisation (with its bass-boost and treble-cut) where the 'quiet' amplifier has become subjectively far worse with an audible rumbling background noise.

Hum pickup of both 50Hz and 100Hz harmonics is also a problem particularly if the head-amp is mains powered. Some manufacturers (for example Ortofon) have managed, through the use of toroidal transformers and plentiful screening, to build the power-supply into the same case as the amplifier section; but most designs use an external power-supply this often being a calculator-style unit on the mains plug. One of the problems often found with mains-powered head amplifiers is that of earthing which, if incorrect, can lead to adudible hum and RF breakthrough. Several arrangements are normally possible (see Fig. 3) and some experimentation may be necessary to obtain the best results.

The installation of battery powered head-amplifiers is rather less critical but these are not without problems. The battery has a finite life and to avoid a rapid run-down it's necessary to remember to switch the unit off after use. Furthermore the use of a common battery for both channels can lead, in some designs, to poor crosstalk and hence poor stereo. The output impedance of the battery depends upon its type and better results are often obtained by using the more expensive and popular 'Mallory' Duracell types.

Some head amplifiers offer a choice of input resistance matchings but those with a fixed input resistance can be changed by soldering a padding resistance into the input plug or by using a matching box such as the QED 26/3. Remember, though, that the reistance can only be reduced not increased and that the following formula applies:

$$R\ \text{INPUT} = \frac{R\ \text{ext.} \times R\ \text{amp}}{R\ \text{ext.} + R\ \text{amp}}$$

where R ext is the extra resistor and R amp is the original amplifier input resistance.

Thus if the head amplifier has 100ohm matching and we add another 100ohms across the input the new input resistance will be:

$$\frac{100 \times 100}{200} = 50\text{ohms}$$

The choice of input matching resistance can be difficult and often the user can only seek advice from the cartridge manufacturer and the maker of the head amplifier. Some combinations seem to be quite tolerant of matching while with others it can be quite critical. Several factors are at work including the maximum power transfer of the signal from the cartridge coils, and the degree of electrical damping provided by the terminating resistor.

CHOOSING & USING A STEP-UP DEVICE

You will need to buy a step-up device if your amplifier does not have an moving-coil input or if the one provided proves to be unsatisfactory. In choosing a suitable unit the factors to be considered include the step-up ratio or gain, the input matching impedance, the noise performance, the convenience of use and the sound quality.

The optimum gain required depends upon both the cartridge's output level and the preamplifier's input sensitivity. For example, consider a preamplifier whose input sensitivity is 3mV and moving-magnet and moving-coil cartridges whose outputs for the same record groove modulation are 3mV and 100 microvolts respectively. Then a step-up device with a gain of 30 will bring the signal up to 3mV and therefore the volume control will remain in the same position. If the step-up device has a gain of only 15 the volume control will have to be advanced to double the gain and so in consequence the signal-to-noise ratio will worsen by 6dB. Suppose we now use a step-up unit with a gain of 120, the signal level will increase to 12mV. As a result the volume control will have to be backed-off and you may find that it is being adjusted in that awkward 'on-off-on' region at the start of the track. Another consequence of this surplus gain is that (in our example) the overload margin has been reduced by some 12dB and so clipping of the signal may occur during loud passages.

As a guide to the relative outputs of the cartridges it is worth taking the Shure V15-III as a typical moving-magnet cartridge with its sensitivity of 1·2mV/cm/sec. A glance at the manufacturer's specification or the better reviews will show the equivalent figure of your intended new cartridge. For example, the sensitivity of the Ortofon MC30 is 0·04mV/cm/sec, so in this case a gain of 30 times will increase the signal level up to that of the Shure cartridge.

The choice of input resistance again depends upon the cartridge and as a general rule it should be greater than the cartridge's own coil resistance. The manufacturer or importer should be able to give this information and the resistance will be found to range from under 1ohm (Audio Note) to over 40ohms (Sony XL55).

If the 40ohm cartridge is loaded by a 10ohm input resistance the signal voltage will be reduced by a factor of 5 which rather defeats the object of having a step-up device! By contrast the signal loss using a 100ohm input resistance is only 0·7. Thus it would seem that the higher the input resistance the better but in practice this isn't always the case.

The noise generated by the step-up device should be as low as possible for, in theory at least, the moving-coil cartridge is capable of higher efficiency and hence, a greater dynamic range than the comparable moving-magnet cartridge. In the absence of understandable specifications (manufacturers can choose many different ways of defining the noise level) the reader can use his ears to make comparisons. With the volume control set for the normal listening level the cartridge can be lifted from the record. The residual noise should then ideally be inaudible.

In terms of user-convenience the transformer scores for it is essentially a fit and forget device. Battery powered head-amplifiers are probably the least convenient because they have to be switched on for every listening session and switched off afterwards, although experience has shown that this latter chore is all to often forgotten. The result is an all too short battery life and I, for one, object strongly to the penal prices charged for batteries.

A & R HA10 Head amplifier

The HA10 is primarily designed to plug into the A & R A60 amplifier from which it derives a power-supply. It can, however, be used separately with the provision of an external supply. It is housed in a small steel case (140 × 97 × 32mm) which is painted eggshell black. There are no controls only a pair of nickel plated phono input sockets (which proved rather prone to tarnishing), an earth terminal, a short (250mm) output lead terminated by a good quality 5 pin DIN plug, and a miniature jack-socket for whenever an external power-supply is used. Inside the HA10 has five small board mounted slide switches (in one block) which can be used to select alternative gains and input loadings.

RESULTS

This unit gave a good all-round performance and was tested using the companion A60 amplifier as a power-source but a separate power supply is available as an optional extra. The distortion and overload margins were fine and the noise level fairly low although there were some ripple components on the output. The hum rejection was about average.

Subjectively the results were good placing this model close to the top end of the units tested.

Merit ★★★★
Value ★★★

Audio Technica AT630 transformer

The AT630 is the smallest of the step-up devices we have reviewed. The transformer is housed in a black painted tube of 30mm diameter and is about 100mm long. The tube is mounted on a neat plastic stand but the AT630 lacks sufficient mass to be stable so it is best for it to be screwed down to the shelf or back of the record deck. At one end there are a pair of gold plated input phono sockets and a ground terminal, while at the other the rather short (400mm) pair of output cables have moulded on gold-plated phono-plugs. Internally this transformer has a triple permalloy shield and is claimed to be suitable for moving-coil cartridges having a coil impedance below 20ohms.

RESULTS

The AT-630 has a voltage gain of 20dB (10 times) and is suitable for low impedance cartridges. It has only a limited signal handling capability at low-frequencies before the distortion starts to rise and the hum rejection is also fairly poor.

Subjectively the sound quality was surprisingly good and so in view of the price this transformer must be seen as good value.

Merit ★★
Value ★★★★

HEAD transformer

The HEAD is quite a large transformer (76 × 76 × 250mm) and heavy with it (3kg). The case appears to be made of aluminium painted black with 'gold coloured' anodising of the two ends. All the connections are on the back panel and use very high quality gold-plated phono sockets which use a split collet to provide a good grip and contact of the centre pin. Three inputs are provided each offering an alternative input impedance and these are marked 4/15/14ohms. There is also a binding post which provides a ground connection.

RESULTS

The transformer gave a very good set of results in the laboratory and no weak areas were revealed.

The listening tests showed the HEAD transformer to be a top-flight performer which seemed to be almost transparent in its operation. This is a very expensive step-up device but can be well recommended.

Merit ★★★★★
Value ★★★

Lentek head amplifier

The Lentek head-amp is now getting rather long in the tooth but it has sold well over the years. Originally designed to match the Entré cartridge it is also recommended as suitable for any other medium output MC cartridge happy with a loading of 100ohms. The Lentek is battery powered and houses a PP3 size cell inside the steel case (150 × 65 × 25mm). This is covered by a piece of aluminium extrusion finish in black crackle paint. The overall result is both smart and neat in appearance.

At the front there is a three position toggle switch having on, off, and test positions, the latter with a spring bias to prevent its continuous use as in this position a LED should light to confirm that the battery is OK. The effect was spoilt on my sample because the LED didn't line-up with the front panel hole.

The Lentek has a pair of gold plated phono socket inputs (no earth terminal) and a rather short (0·5 metre) output cable terminated by gold plated phono plugs.

RESULTS

The voltage gain was slightly dependent upon the source impedance a figure of 29dB being recorded for a 39ohm source. The distortion was a little high with a 5mV input signal but this was mainly a reflection of the headroom which was fairly limited clipping occurring at about 12mV input. The figure for stereo separation was fine but the hum rejection was poor so care must be taken in the siting of this model.

Subjectively the sound quality was good although there was some colouration heard which made the overall balance rather 'warm'. Despite a fairly indifferent noise figure, background noise never intruded.

Merit ★★★
Value ★★★

Mayware transformer

The Mayware transformer is housed in a fairly small (100×60×55mm) steel box painted black. There is one set of input and output sockets which are gold plated phono types. There is also a miniature binding post which provides a ground connection to the case. The Mayware transformer is recommended for cartridges which have an impedance of 2 to 40ohms. Obviously by having a fixed turns ratio the gain will be affected by the transformer matching.

RESULTS

The Mayware proved to be very well screened and as a result there was very little hum pickup. Otherwise the results were about average although when fed from higher impedance cartridges there was some loss of high frequencies.

Subjectively the Mayware proved to be quite neutral in the midband but bass seemed to lose some of its solidity when used with the Asak but this was not apparent when the Dynavector "Ruby" was used.

Merit ★★★
Value ★★★

Ortofon MCA-10 head amplifier

The MCA-10 is a low cost model introduced by Ortofon which has virtually replaced the expensive MCA-76. The basic case is moulded in grey plastic but is housed in a black steel sleeve which provides some screening (140 × 80 × 40mm). The MCA-10 is battery-powered and two small (IEC LR 14) 1½ volt cells fit neatly into a removable drawer which slides out of the front panel. Also at the front are the on/off switch and a miniature battery meter with green and red segments. The power switch also acts as a signal bypass so that when the head amplifier is off the signal is wired straight through from input to output.

At the back there are ordinary nickel plated input and output phono sockets.

The input impedance is a claimed 11ohms but the gain is dependent on the cartridge resistance a figure of 33dB for 3ohms being quoted.

RESULTS

Whilst the MCA-10 is undoubtedly very quiet it can be used only with low output cartridges because output clipping occurs at an input level of only 7mV. Excellent figures were recorded for hum rejection.

Subjectively the MCA-10 was adjudged to be a fine step-up device definitely on a par with the more expensive Trio unit. In all this head amp represents good value and can be recommended.

Merit ★★★★
Value ★★★★★

Ortofon T-30 transformer

This larger Ortofon transformer is of a similar styling to the T-20 with the same black steel case (80 × 40 × 145mm) and anodised front panel. The input and output phono sockets are, however, gold-plated and there is also a ground terminal which is also gold-plated. On the front panel there is a 6-position rotary switch which can be used to bypass the transformer or to switch-in alternative ratios to provide matching for cartridge impedances of 3/6/12/24/48ohms.

RESULTS

We found little to criticise about this transformer except its poor shielding to magnetic hum fields. Care should, therefore, be taken to position the T-30 well away from amplifier transformers and turntable motors etc.

Subjectively the sound quality was very good although it was felt that the bass was less coherent and well defined than when using the HEAD transformer.

Merit ★★★
Value ★★★

PS Audio head amplifier

This particular head-amplifier is an expensive mains-powered unit made in the USA. The main case (200 × 110 × 40mm) is of steel given a satin chrome plating. Over the top is a removable black anodised aluminium cover which gives the unit an untidy overall appearance. At the front there is a giant-size power-switch; a supply indicator LED and a cable outlet to the moulded calculator style mains power supply. Around the back there is a binding post grounded to the case and good quality nickel plated phono sockets for the inputs and outputs. The input impedance is claimed to be 30ohms although this can be reduced by inserting resistors (supplied) into sockets on the PCB. The voltage gain is fixed at 27·5dB.

RESULTS

The voltage gain proved to be about 27dB but with a measured input impedance of below 30ohms this preamp is only ideal with low resistance cartridges. The results for overload and distortion were fine but the hum rejection was fairly poor.

Subjectively this head amp was found to be a very successful design which worked well with a variety of test cartridges.

Merit ★★★
Value ★★

Rogers MCP100 head amplifier

The MCP100 is supplied as a (free) standard accessory with the A100 amplifier but is also sold separately. It is a battery powered unit using a single PP3 size cell in a steel case (105 × 105 × 55mm) finished in black with white lettering. On the front panel there is a rocker switch with on, off and battery test positions the latter having a spring bias to prevent battery drain by the LED indicator (a battery life of about 400 hours is claimed).

At the back there is a ground terminal and the input and output sockets all of which are gold plated. Accessible from underneath are six miniature slide switches on each channel. These give a choice of ×14 or ×30 gain; input loading resistors of 33, 43, 114, and 470ohms; input shunt capacitance of 10, 22, 32, 68, 78, 90, and 100nF although in practice such 'fine tuning' of capacitance is unlikely to be of any significant benefit.

RESULTS

The voltage gain was measured to be as the manufacturer's specification. Hum rejection was fair as was the harmonic distortion but the input overload headroom was limited clipping occurring (on the × 30 gain setting) at a little over 10mV input eg, 26dB over 0·5mV.

The noise level was moderately low and unobtrusive in use and the stereo separation was fine.

Some colouration was noted in the listening tests and a tendency towards smoothness at the top-end which, though pleasing, was thought to be inaccurate.

Merit ★★★
Merit ★★

Trio KHA-50 head amplifier

The KHA-50 is a smart little moving-coil head amplifier from Trio's 'High Speed' series. The steel case is fairly large (180 × 120 × 45mm) and is covered by a piece of brushed aluminium extrusion which gives a high quality finish. At the front there is a push-button for power and a green LED indicator. At the back there are gold-plated phono sockets for the inputs and outputs as well as an earth terminal. There is also a plug-in cable which carries the 12volt DC supply from a small calculator-type power supply which has an internal fuse.

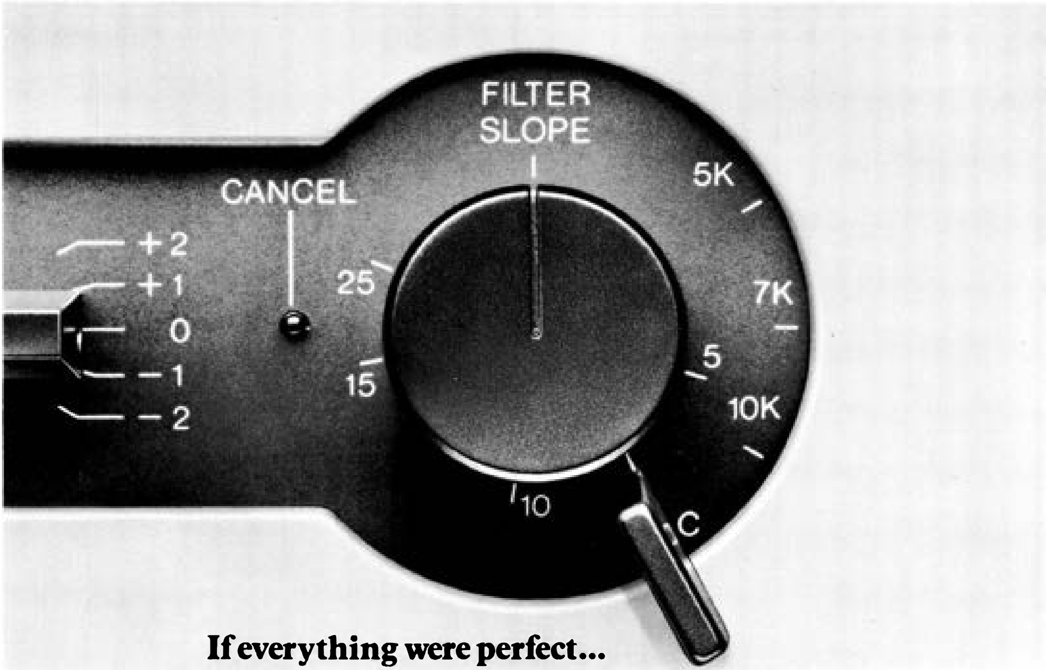
No input resistance has been quoted but the channel voltage gain is a claimed ×25 (28dB). Trio also claim a rather extended frequency response; −3dB at 2MHz!

RESULTS

This head-amp proved to be the quietest yet tested, had very low distortion, a flat but extended response and was fairly uncritical of source impedance. The hum rejection was about average whilst the voltage gain measured at 28·5dB.

Subjectively the Trio was well liked with a clean, open sound quality enhanced by an apparent absence of low-frequency noise.

Merit ★★★★★
Value ★★★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MOVING COIL STEP-UP DEVICES

		A&R HA-10 (RI selected)	Audio Technica AT630	HEAD (15ohm setting)	Lentek	Mayware	Ortofon MCA-10	Ortofon T30 (3ohm setting)	PS Audio	Rogers (×30 setting 114ohms)	Trio
GAIN											
2Ω Source	(dB)	29·2	20·4	21·7	26·6	26·9	34·1	28·8	27·5	29·6	28·4
10Ω Source	(dB)	29·3	20·7	21·9	28·4	27·2	34·3	28·9	27·7	30·0	28·6
39Ω Source	(dB)	29·3	20·8	21·8	29·0	27·2	34·3	28·9	27·7	30·0	28·6
Maximum I/P Voltage for 0·5% THD	20Hz	85·0mV	See Plots	See Plots	12·5mV	See Plots	7·2mV	See Plots	225mV	10·7mV	84·0mV
	1kHz	85·3mV	> 4 volts	> 4 volts	13·0mV	> 4 volts	7·2mV	> 4 volts	227mV	10·7mV	84·0mV
	20kHz	84·9mV			13·3mV		7·2mV		220mV	10·7mV	83·7mV
Input Resistance	Ohms	90	27	(4) 6·1 (15) 23·3 (40) 52·5	94	4·3	10·8	(3) 3·5 (6) 6·4 (12) 13·2 (24) 25·7 (48) 48·8	23		89·9
Hum Rejection 50Hz	(dB)	41	41	63	39	63	61	35	43	45	47
150Hz	(dB)	42	49	-	53	65	66	-	44	47	50
Harmonic Distortion 1kHz 5m I/P dB 2nd Harmonic		−81	−101*	−105*	−66	−89	−53	−97	−64	−77	−89
dB 3rd Harmonic		−92	−100	−104*	−81	−91	−52	−102*	−99	−76	−92
Stereo Separation	1kHz	78	below noise	below noise	>80dB	below noise	>80dB	below noise	79	>80	>80dB
Signal-to-Noise Ratio Equalisation ref 0·5mV (dB) after RIAA	CCIR/ARM	79	-	-	76	-	82	-	77	75	86
	Unweighted 20Hz to 30kHz	56*	-	-	70	-	73	-	42*	70	65
Noise Figure	(dB)	-	1·9	<1	-	1·6	-	1·3	-	-	-

Conclusions

The results of the laboratory tests are shown in the table and, for the large part, are self-explanatory. The hum rejection test is really a test of the effectiveness of the shielding in minimising the pickup of hum (50Hz) com ponents from such sources as amplifier power transformers.

Of the products tested there was a slight preference in the listening tests for the transformers. The most preferred models were the HEAD and the Ortofon but the differences were smaller than expected. It should be remembered that in the mid and high frequency bands the transformer is effectively two coils coupled only by air, the core material only being necessary to improve the performances at low frequencies. The audible results of different designs were mainly to be heard in the low-frequency reproduction which in some cases was lumpy in response terms or lacked definition. Some of the transformers imposed a load on the MC cartridge which is not constant with frequency and as a result there was some distortion audible as a grittiness at high frequencies.

Good results were obtained with the Trio, A & R, and Ortofon headamps, but it was notable just how great the audible differences were between the designs. In design they are mostly much simpler than the equivalent MC stage built into good quality amplifiers. This is a result of higher cost of building a separate unit.

However our tests did show that there can be appreciable interfacing effects between the cartridge/step-up device/preamplifier and we cannot predict that our liking for the sound quality of one transformer design will be as strong when another cartridge is used. So we cannot stress too highly that the intended combination should be tried first before buying.

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