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NEW DEVELOPMENTS IN AUDIO RECORDING AND REPRODUCING CIRCUITRY FOR ANALOGUE TAPE RECORDERS

STUDER INTERNATIONAL AG
Regensdorf / Zürich, Switzerland

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1. INTRODUCTION

- In any Audio Tape Recorder, the signal to be recorded and then played, has to be processed adequately to permit its conversion from an electrical/time dependant value to a magnetic / space dependant state, and vice-versa.

- The purpose of this paper is to point-out where significant improvements have been made in signal processing over the past years and to present state of the art solutions.

- According to diagram (1), the following points will be discussed: Audio input stage, record amplifier and bias circuitry, reproduce amplifier, output stage and parameters programming.

- Most of the solutions described are used in the new STUDER A810 Compact Tape Recorder.

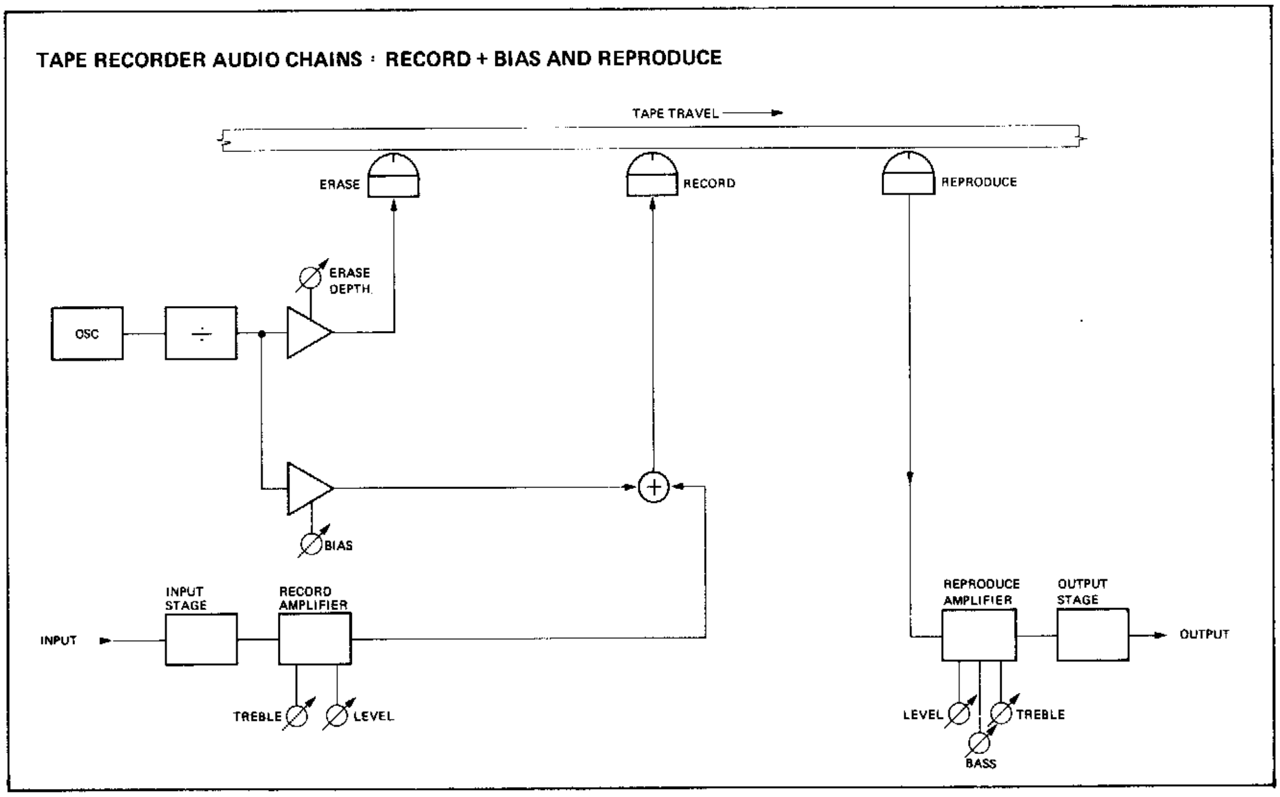


figure 1

2. THE INPUT STAGE

- In professional audio field, there is no need of strong mathematic developments, nor years of experience to be convinced that balanced lines and associated balanced input and balanced output stages have to be used if any interferences have to be removed from transmission lines without problems.

- However, the way balancing is achieved, and how good this balancing, are influencing greatly the overall performance of a balanced transmission line.

- There are basically two ways to design a fully balanced line input stage:

  a) Transformer Balancing
  b) Electronic Balancing

- The following table shows the advantage and the drawbacks of both balancing modes.

Parameter (line output amplifiers)	Transformer	Transformerless
- Frequency response	2	3
- Intermodulation, IMD	2	3
- Harmonic distortion, THD	2	3
- Transient intermodulation, TIM	3	2
- Phase modulation distortion ΦMD	1	3
- Phase response	3	3
- Radiated stray-field	2	3
- Common mode rejection Ratio CMRR	3	2
- Safety	3	2
- Common mode rejection range	3	2

NB: 3: very good 2: average 1: poor

It is interesting to note that these considerations are also valid for a line output stage with only minor alterations.

- The conclusion to be drawn from this table is immediate:

It is not possible to state definitively and as a general rule that electronic balancing method is better or worse than transformer balancing method. But, instead, it is easy to define which method is best suited to a given application.

E.G.

- Transformers will be best suited when recording on location with mobile units or O.B. Vans, or generally in radio studios located near transmitting site. They are also a best solution for nearly any non-fixed installations, like concert P.A. systems, film and T.V. sound recording outside of studios or when transmission over relatively long lines is required.

- Electronic balancing will be best suited for disc recording studios, specially where a great number of sound processing equipment (special effects) is required or when recording classical music. Of course, in this case, transformerless condensator microphones associated with transformerless microphone input stages on mixing console are of prime importance.

- The following diagram (2) shows the basic principle of a transformerless balanced line input stage, however, this design is not perfect because if perfect balancing is achieved, then C.M.R.R. is poor or vice-versa.

BALANCED TRANSFORMERLESS LINE INPUT STAGE

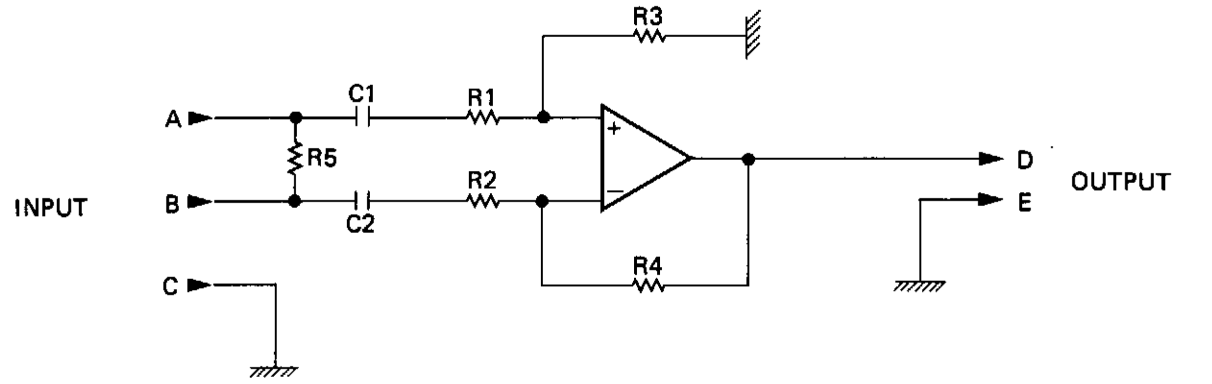


figure 2

The balancing condition is:

Gain A = Minus Gain B

The maximum CMRR condition is:

RA = RB

But: RA = R1 + R3

And RB = R2 + f (VA)

where f (VA) represents a resistance varying according to the input voltage VA.

- It is then easy to show mathematically that these two conditions cannot be satisfied simultaneously.

- Fig. (3) shows a very good example of a simple but effective design:

BALANCED AND FLOATING TRANSFORMERLESS LINE INPUT STAGE; 2 IC

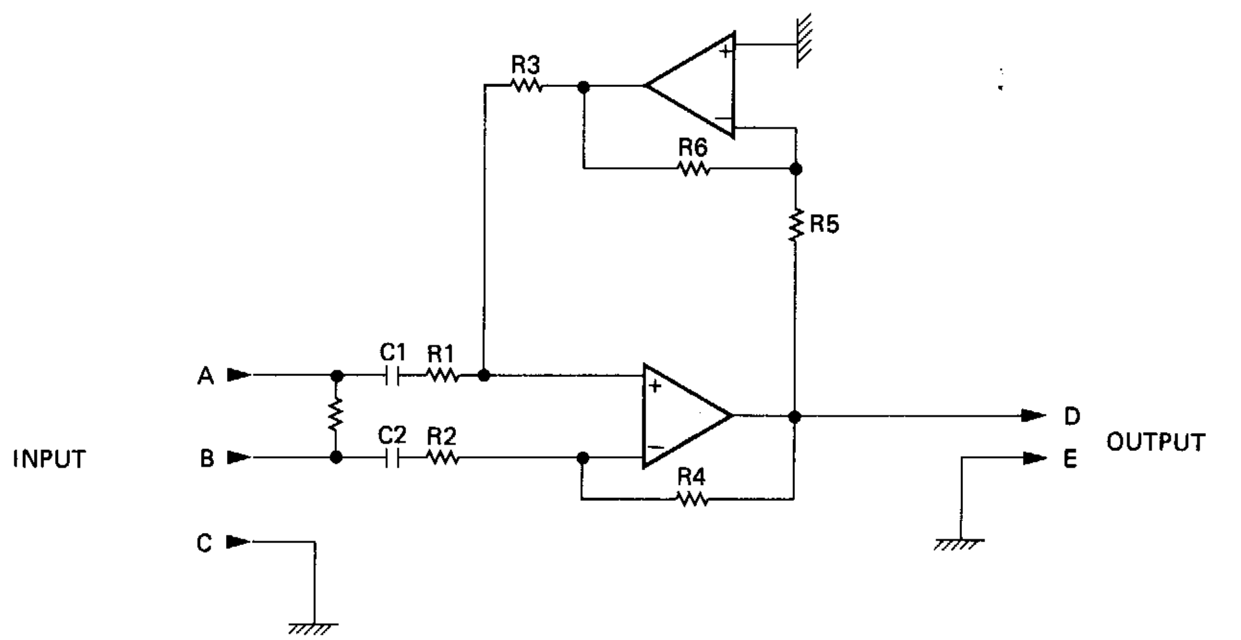


figure 3

$GA = \frac{R3}{R1}$ and $GB = \frac{RY}{R2}$ (if: R6 = R5)

RA = R1 and RB = R2

However, the propagation time (delay) of the audio signal through IC-2 can induce some oscillations at high frequency and thus the bandwith will be reduced if filtering has to be used to avoid unwanted oscillations.

- Fig. (4) shows a design where band limiting problems are not critical but at the expense of an additional I.C. This is the solution which STUDER retained.

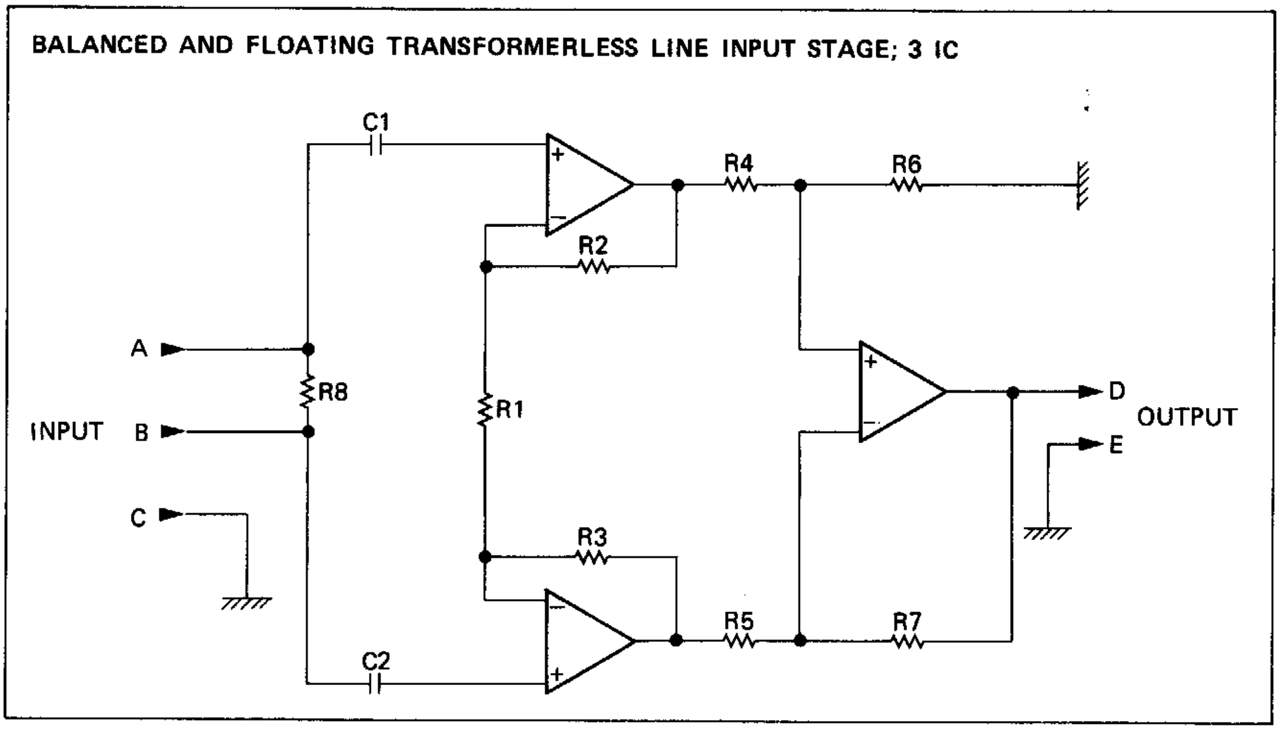


figure 4

- Usually, transformers have been used in line input stages, more or less in the way described on Fig. (5). However, it is easy to show that the THD generated by transformers is a direct by-product of the internal magnetic induction B. In other words: $\vec{B} = \vec{0} \Rightarrow$ No more THD (Core Material Saturation).

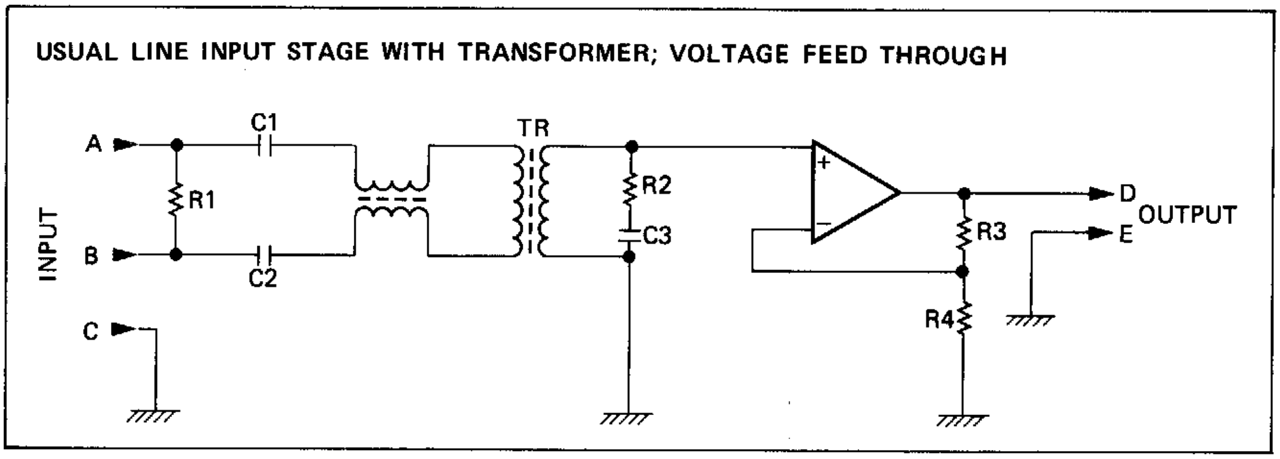


figure 5

- In a transformer, the output voltage is given by: $Vo = - k \frac{dB}{dt}$; so, if we short circuit the output, and that Rs is zero, then $Vo = 0 \Rightarrow B = 0$.

- Actually, transformers do have a secondary internal resistance Rs which is not zero. If we have a look at Fig. (6), this means that the input resistance of the following stage has to be negative Ri = - Rs .

ELECTRICAL EQUIVALENT CIRCUITRY OF AN AUDIO TRANSFORMER

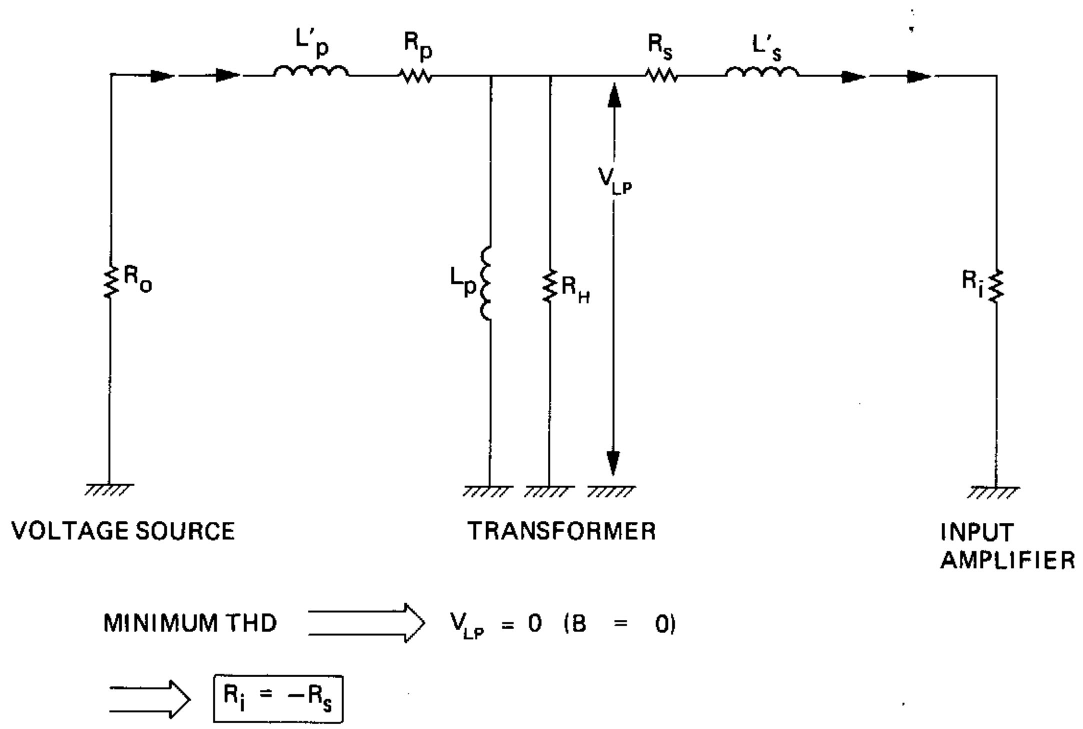


figure 6

- Fig. (7) shows a practical example of a line input stage where the transformer secondary coil is loaded by an amplifier having a negative input impedance. This is the solution retained by STUDER, it provides very low THD at all frequencies combined with usual balancing quality.

LINE INPUT STAGE WITH TRANSFORMER; CURRENT FEED THROUGH

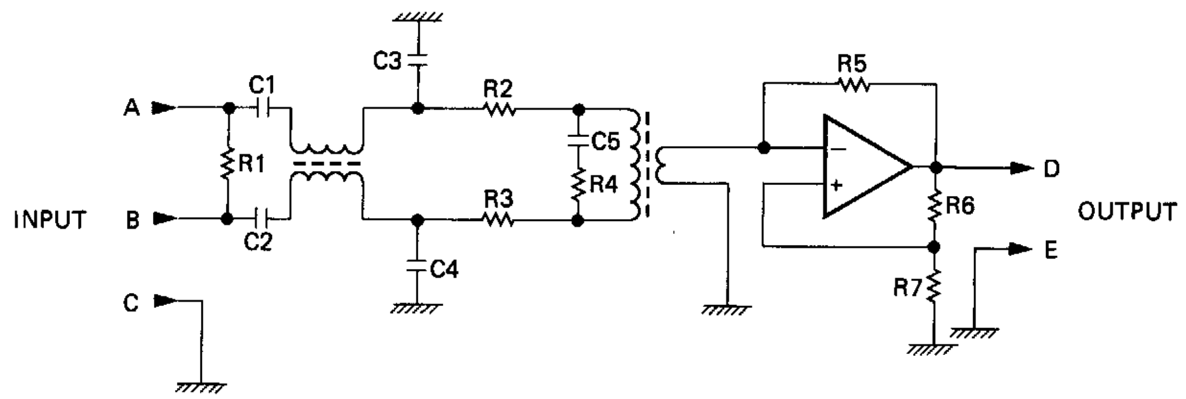
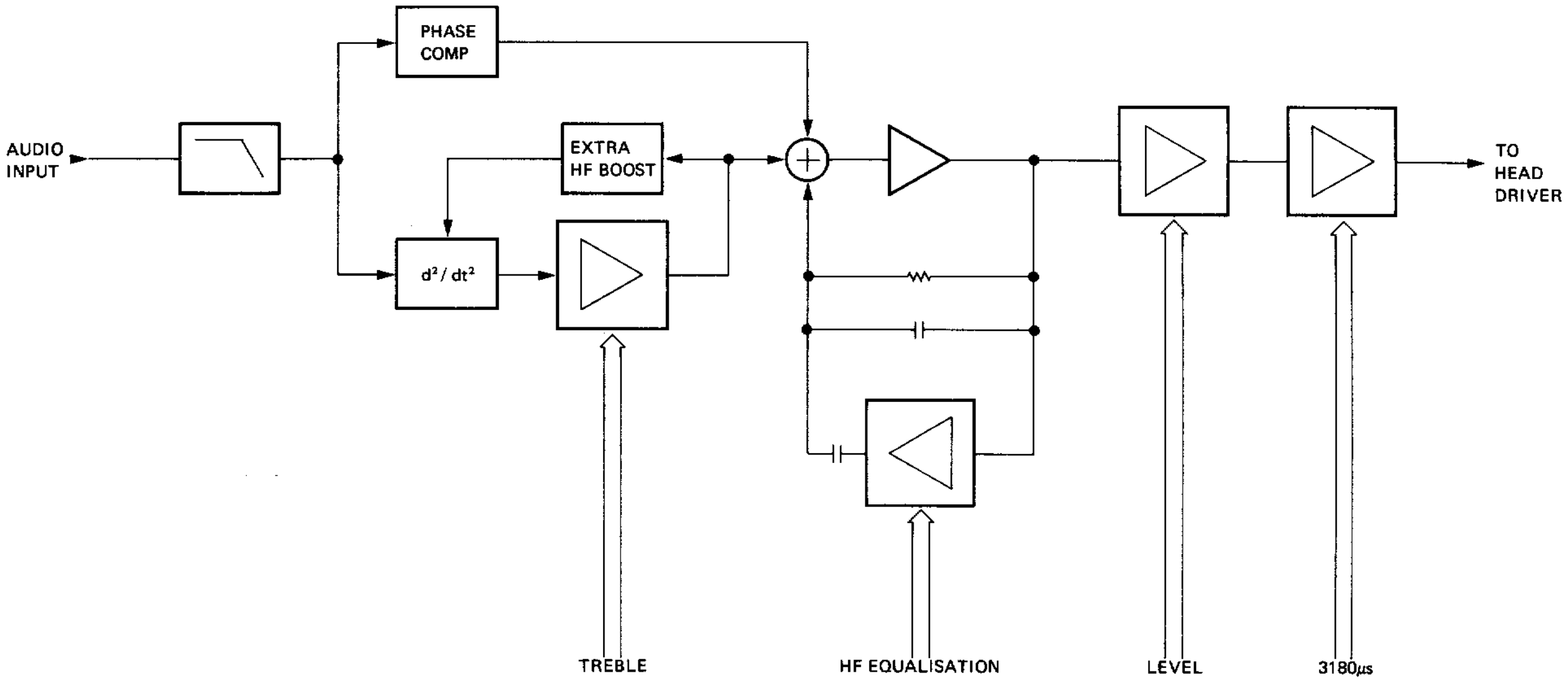


figure 7

RECORD AMPLIFIER : BLOCK DIAGRAMM



3. RECORD AMPLIFIER AND BIAS CIRCUITRY

- The recording process is not simple because the magnetisation of a tape is a highly frequency dependant function. This function is complex, and the solutions used for its approximation are greatly influencing the signal quality, specially when phase / frequency behaviour is analysed.

- Fig. (8) shows a modern recording amplifier block diagram.

- First, there is a low-pass filter removing any high frequency components (corner frequency 25 to 30 kHz) That could induce some objectionable mutual biasing.

- Then the signal is routed via an all-pass filter to an amplifier having a feed-back which is frequency dependant. The corner frequency of the feed-back loop is selectable by mean of a programmable resistor dividing network. The result is a -6dB / octave slope occuring above a corner frequency corresponding to the time constant of the required equalisation, i.e: 120, 70, 50, 35 or 17.5 µs.

- However, high frequencies are resulting in short wavelengths on the tape, and these are difficult to be reproduced due to the self erasing phenomenon which occurs at the surface of the tape.

- A relatively good approximation to compensate for this sensitivity loss is to use at first a +12dB / octave boost over a frequency which is dependant of the tape type and the recording speed, and then a further extra +6dB / octave boost at about twice this frequency.

- The phase distortion (2 x + 90^o) generated by the d2 / dt2 circuit (+12dB / octave) is compensated by the all pass filter [phase compensation].

- The remaining + 90^o phase distortion brought by the extra-boost stage are occuring at a frequency where compensation do not bring a significant improvement.

- A programmable 8-bit (256 steps) amplifier ensures fine recording treble adjustment.

- Another programmable 8-bit amplifier ensures fine recording level adjustment.

- Finally, an ordinary electronically switched amplifier enables the NAB 3'180 µs time constant when required.

- Fig. (9) shows the effects of all these adjustments on the transfer function of the recording amplifier.

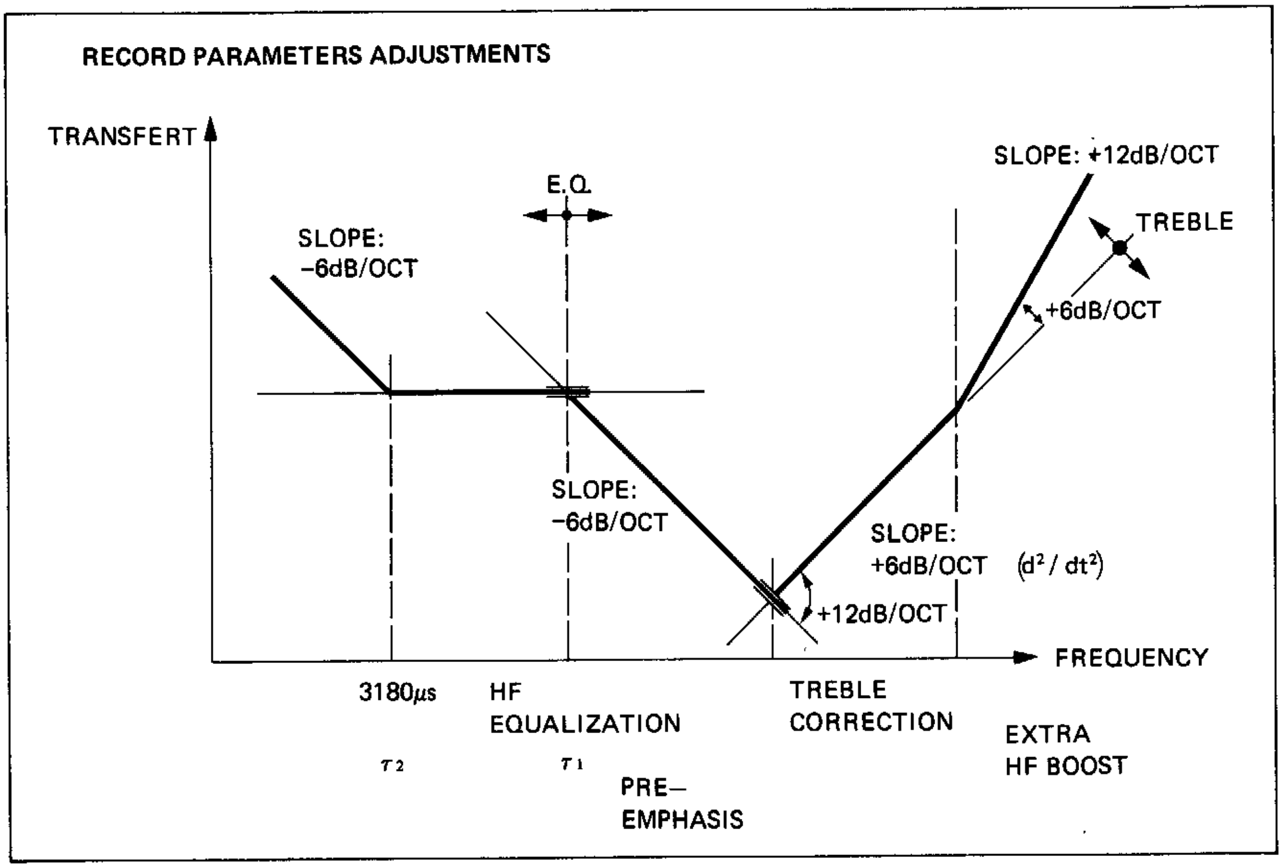


figure 9

- Fig. (10) shows a record head driving amplifier having a current output which is not dependant of its load. In other words, io = f (Vi); io = g (ZL). This is achieved by the resistor network R3, R4, R5 which generates a feed-back voltage proportional to the current output of the amplifier.

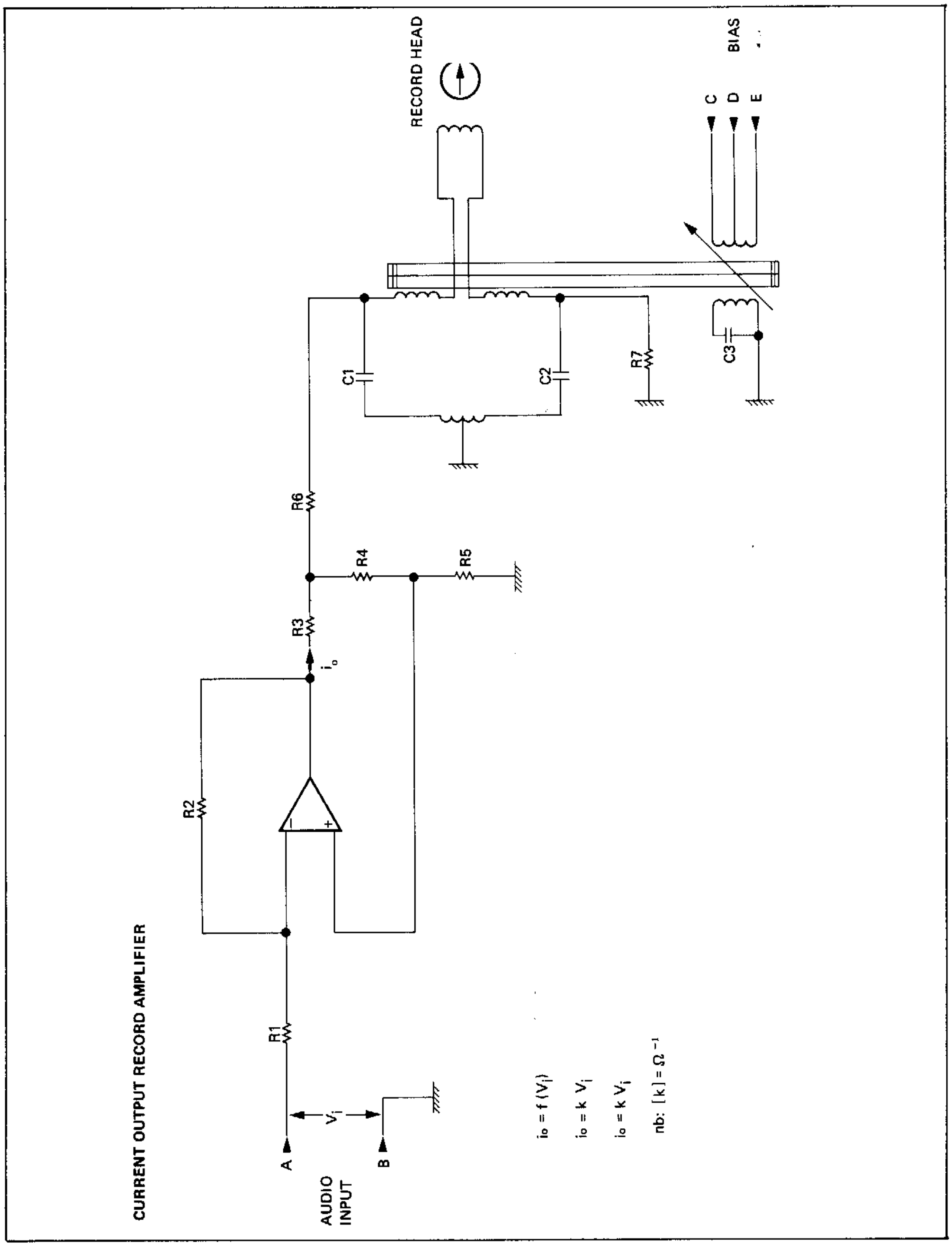


figure 10

- The bias circuitry shown on Fig. (11) is interesting because
  - a) it is programmable and
  - b) it can be extended without problems to multitrack tape recorders.

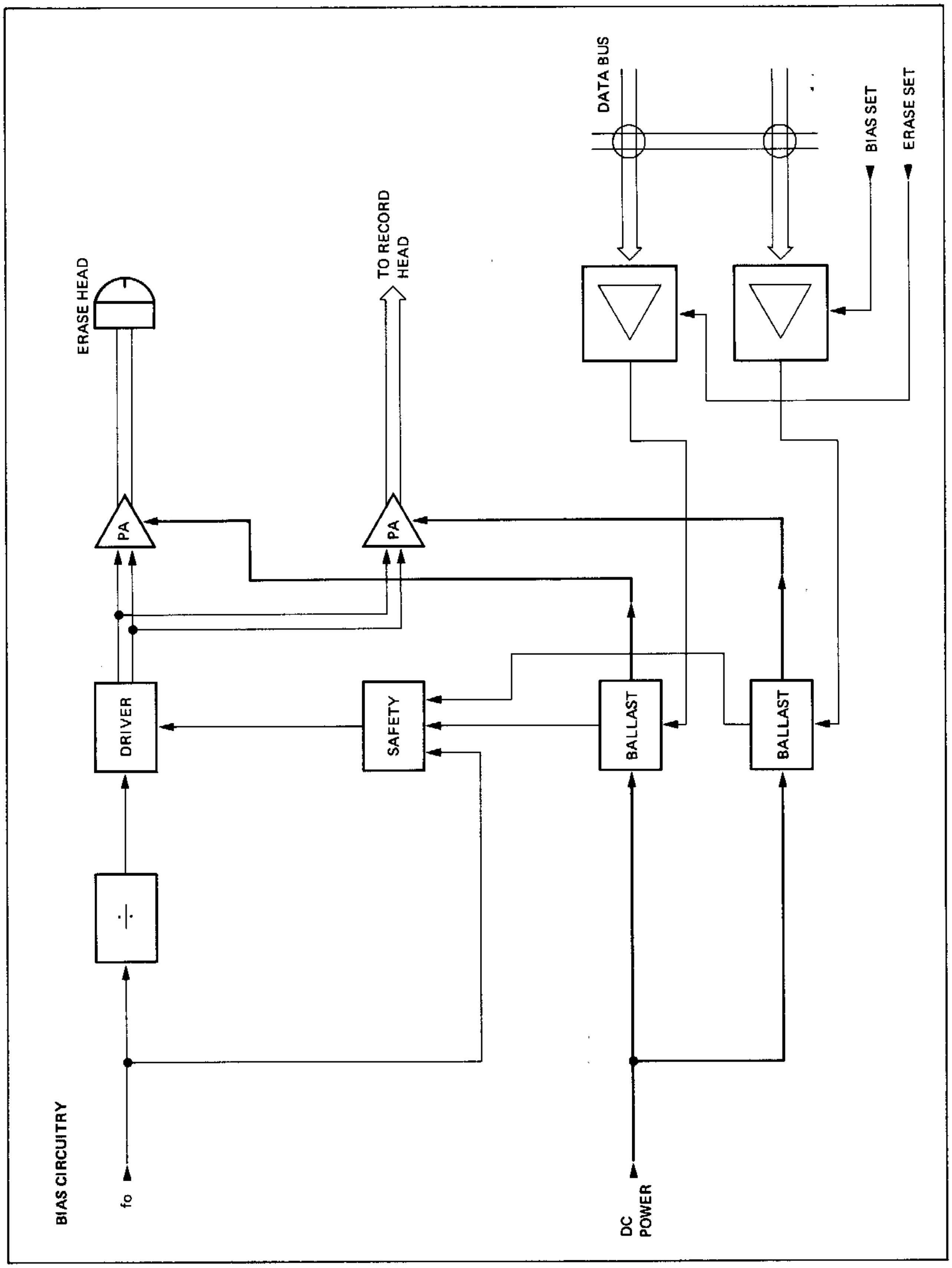


figure 11

- The reference frequency is at first devided down to the appropriate value. Then it is shaped and cleaned by the driver to yield a symmetrical square wave. This driver can be switched off if either reference frequency or power control of the P.A.'s is failing.

- Then there are some power amplifiers having an output level directly proportional to their DC voltage supply output signal being a pure sinusoid.

- These DC voltages are supplied by regulating stages (ballast) which are controlled by programmable attenuators. In this case 8-bits (256 steps) for biasing and 4-bits (16 steps) for erasing.

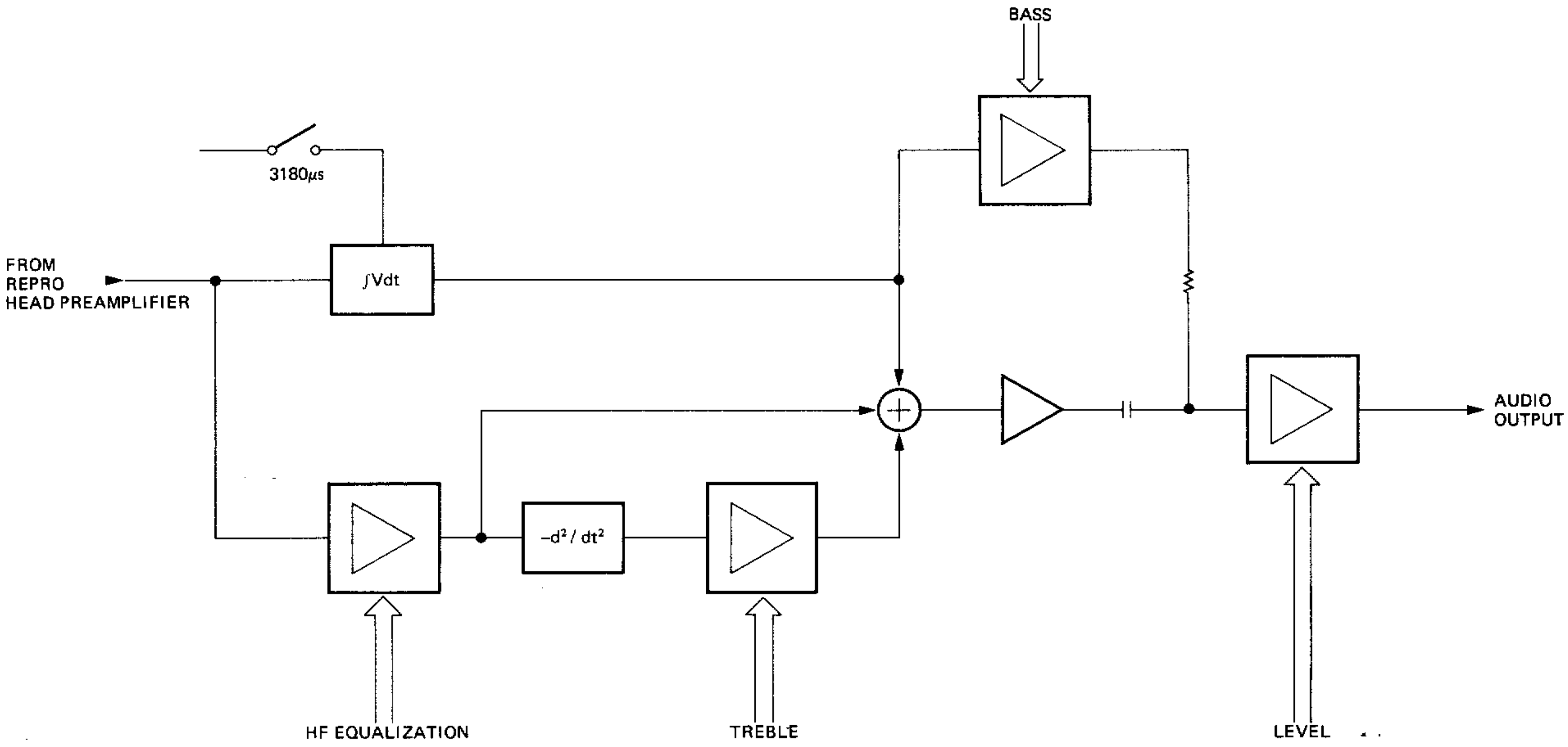
4. REPRODUCE AMPLIFIER

- Fig. (12) shows the block-diagram of the reproduce amplifier of the STUDER A810 compact tape recorder.

- At first, a global integration giving a 6dB / oct. attenuation over the whole frequency spectrum is provided (S v dt.). However, a switchable time constant of 3'180 µs in the integration amplifier allows linear or NAB low frequency de-emphasis.

figure 12

REPRODUCE AMPLIFIER : BLOCK DIAGRAMM



- Via another branch, the signal level is adjusted by mean of a programmable amplifier which in this way sets the high frequency equalisation time constant (see Fig. 13).

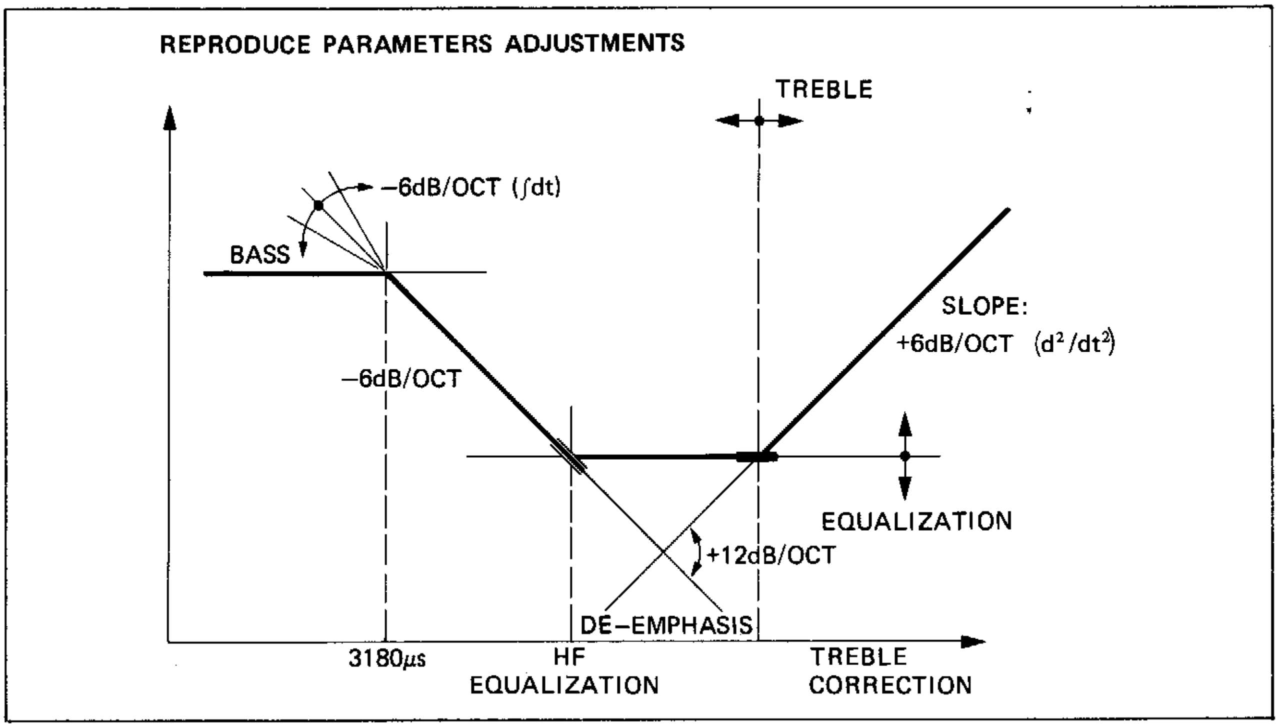


figure 13

- The stage (d2 / dt2) is approximating the head gap loss function ($\frac{\sin x}{x}$) when the deviation is not too important. A higher order derivation is not desirable because of phase distortion problems and noise level in the high frequency part of the spectrum. Treble correction is then adjusted by a programmable amplifier.

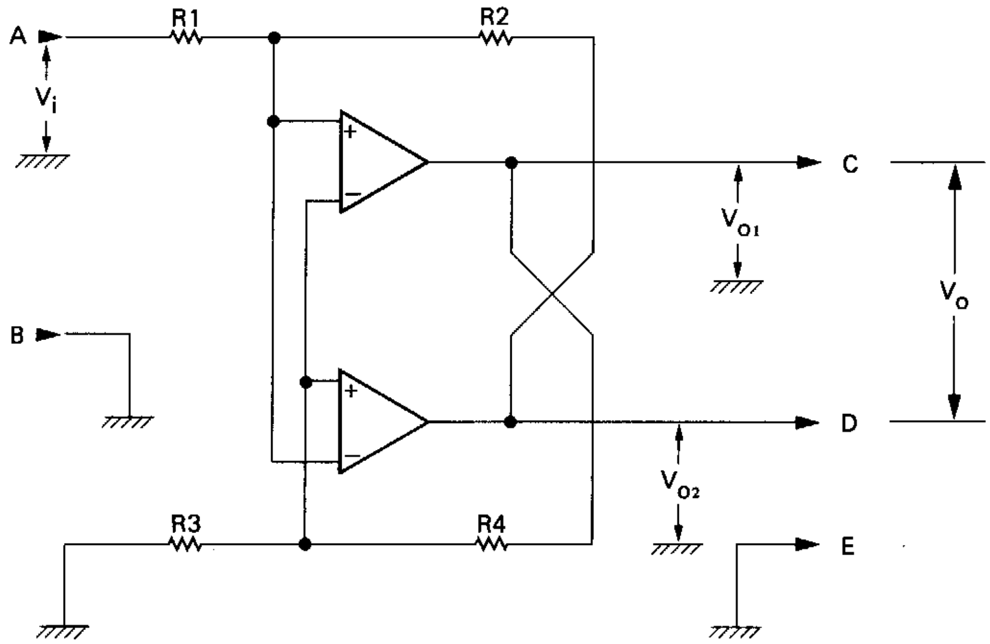
- Bass correction is provided by re-injection of LF signal coming out directly from the integrating stage.

- At the end, the overall signal level can be adjusted also by a programmable amplifier.

5. OUTPUT STAGE

- The line output stage shown on diagram (14) is transformerless, balanced and floating; it is based on a bridge configuration. The practical realisation contains naturally some more components to ensure perfect output symmetry as well as unconditional stability.

TRANSFORMERLESS BALANCED AND FLOATING LINE OUTPUT STAGE



$A \simeq \frac{R2}{R1}$; $\frac{R2}{R1} \overset{!}{=} \frac{R4}{R3}$

figure 14

- On Fig. (15), a wide band transformer line output stage is described; at frequency under 20 to 25 kHz, the amplifier feed-back signal comes (via R2 and a low-pass filter) from a special winding which acts as a separate secondary coil. At higher frequency, the phase difference between the primary and secondary windings of the transformer is too big to avoid feed-back induced oscillations, so, in this case, feed-back signal is picked-up directly from the transformer primary winding via a high pass filter and R3.

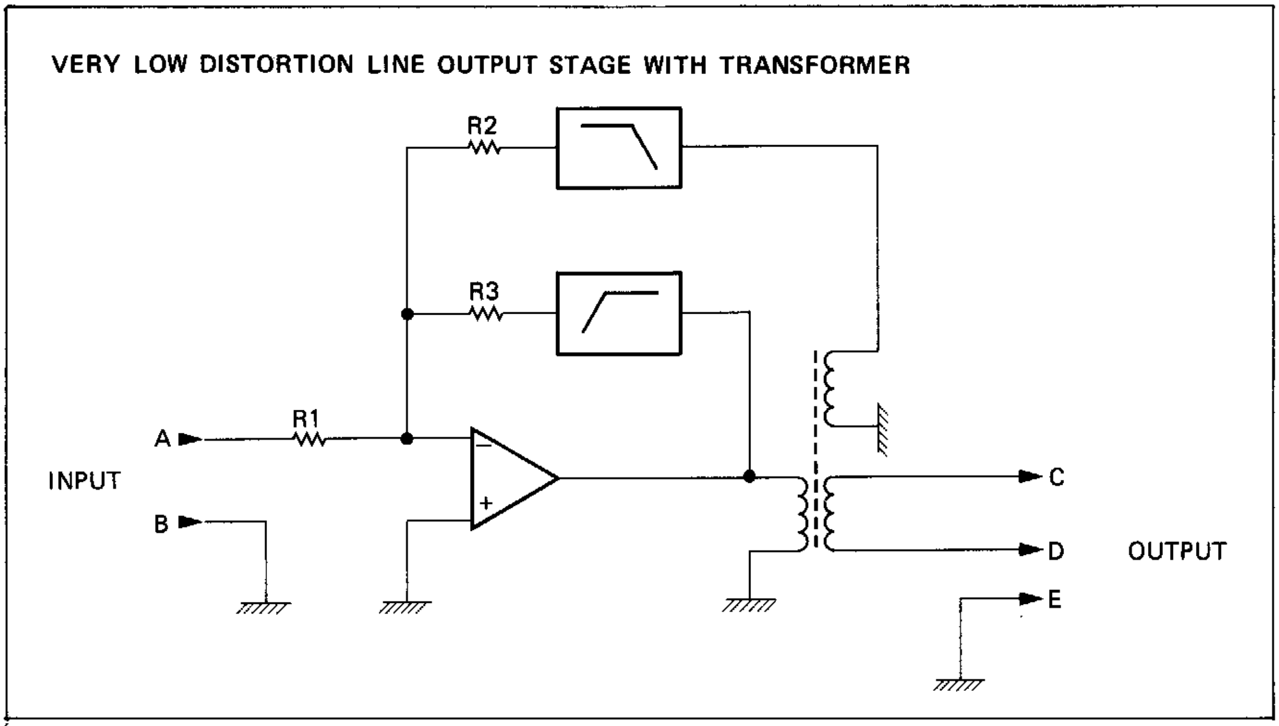


figure 15

6. PARAMETERS PROGRAMMING

- A modern professional tape recorder, even compact or of moderate price, should be able to handle different kind of tapes at different speeds and with different equalisations without complex or endless adjustments.

- So, we thought that parameters programming was a really effective and elegant solution, if not the only one when considering todays requirements.

Inventary of status:

- Equalisation	:	2	(NAB and IEC)
- Speed	:	4	(3,75, 7,5, 15 and 30 ips.)
- Tape Type	:	2	
- Operating level	:	4	(0dBm, + 4dBm, + 8dBm and + 10dBm)
Total Combination	:	64	Status.

- The Fig. (16) shows the principle of our parameters programming system:

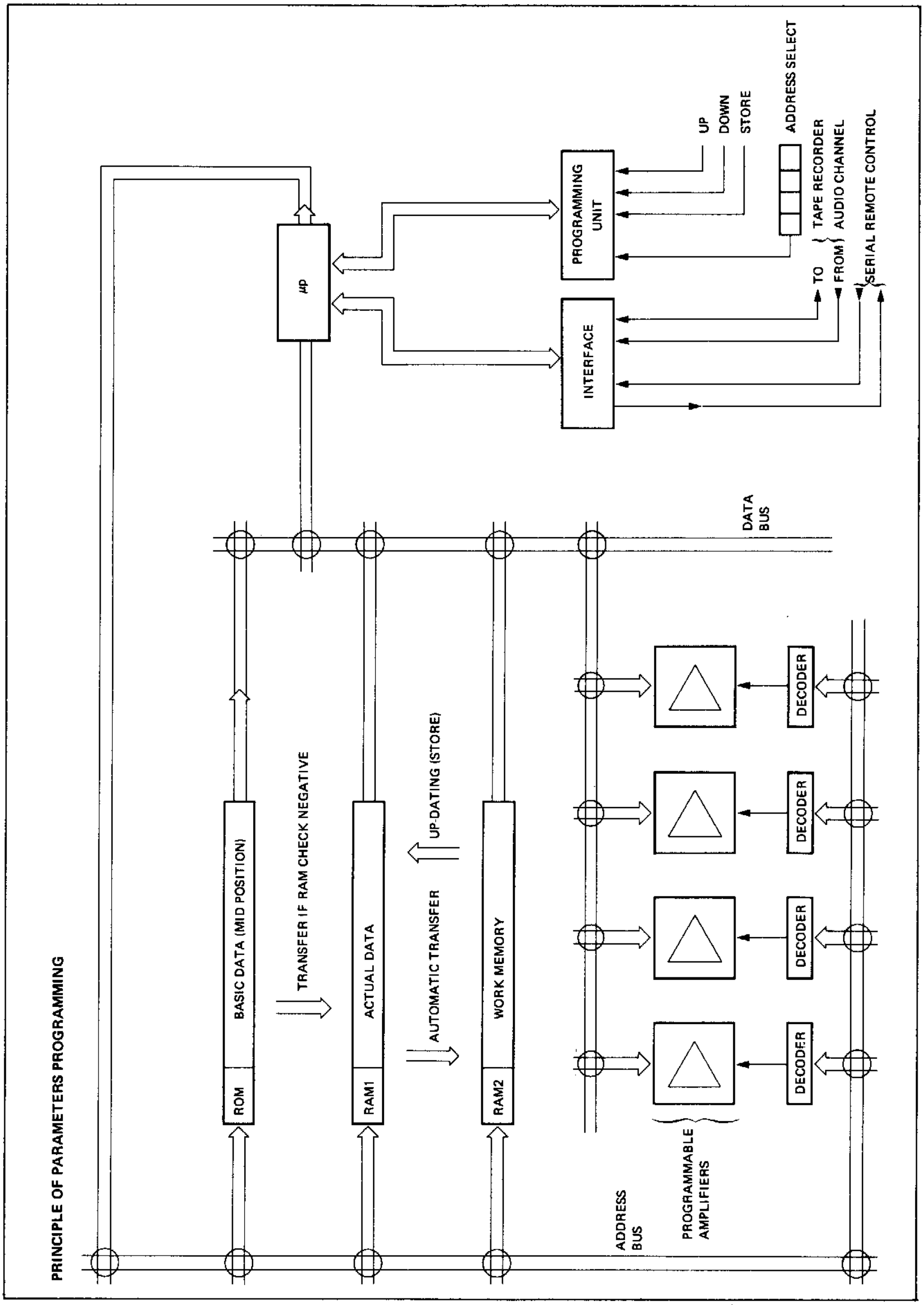


figure 16

a) The basic data is contained in a ROM which sets the mid position and the limits of each parameter adjustment range.

b) Periodically, or after some specific operations, the data stored in RAM-1 is checked, if this test reveals faulty, then the basic data is automatically transfered in RAM-1.

c) The RAM-1 is containing the actual data, i.e: The data which is taken into consideration by the micro-processor to set automatically the programmable amplifiers (or attenuators) values.

d) The actual data bank (RAM-1) can be up-dated by the user if the store function is activated.

e) A working memory (RAM-2) is used for manual parameter programming by mean of the programming unit.

f) A serial interface with adjustable baud rate allows programm transfer from or to an external source, and permits also serial remote control via the STUDER data bus.

g) The programmable amplifiers are permanently connected to the data bus and they receive the set instruction via their respective address decoders.

- A typical programmed controlled amplifier is shown on Fig. (17). It is interesting to note that the external components R1, R2 and R3 are setting the operating limits of the stage: Amplifier or attenuator only, or combined amplifier / attenuator.

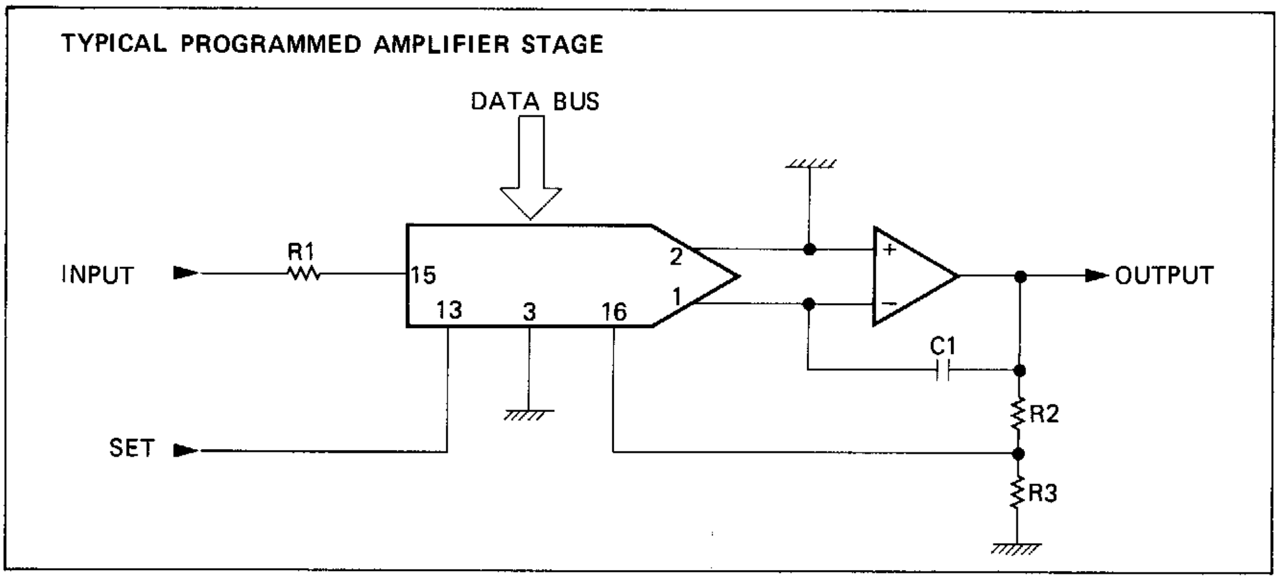


figure 17

7. CONCLUSION

- A tape recorder designed and built with high quality audio circuits and full parameters programmation capability exhibits no quality loss despite its versatility, and reliability is expected to increase over conventional systems thanks to micro-processor systematic check of all key functions.