REPORT 795-3

# TRANSMISSION OF TWO OR MORE SOUND PROGRAMMES OR INFORMATION CHANNELS IN TELEVISION

(Question 47/10, Study Programme 47A/10)

(1978-1982-1986-1990)

## 1. Introduction

High quality multi-channel sound has been shown to be an asset to television programme transmissions for stereophonic sound reproduction and for separate sound channels. Studies conducted since 1959 have led to the development of several different methods of adding additional sound channels and data transmissions to television systems in use throughout the world. Some of these systems have been implemented with two being recommended, see Recommendation 707 and two being proposed in Appendices I and II to Annex I to this report.

A system for transmission of stereophonic or two separate sound signals using two sound carriers, "the two-sound carrier FM system", is in regular service as described in section 3. This service began in the Federal Republic of Germany in 1981 and has also been introduced in other countries, e.g. Australia, Italy, Netherlands and in the Republic of Korea.

The NICAM-728 digital system, where multi-channel sound and data are carried on a digitally modulated carrier additional to the analogue sound carrier has been studied in the United Kingdom and a number of other countries, and by the EBU. This system was put into regular service in Denmark, Finland and Sweden in 1988 and in New Zealand and the United Kingdom in 1989. The system is described in section 4. Studies are proceeding in France on the adoption of the digital system to television system L [CCIR, 1986-90a].

In 1978, an FM-FM system of multiplexing two sound channels on the single sound carrier of television system M was put into service in Japan. Stereophonic sound or two separate sound channels may be provided by this system as described in section 5. Work is under way to implement data services, for example facsimile broadcasting to provide additional text services.

In North America, the United States and Canada have implemented the BTSC* FM-AM multiplex system of multi-channel sound with television broadcasting beginning in 1984. The BTSC system permits the simultaneous transmission of

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* Broadcast Television Systems Committee of the Electronic Industries Association, which adopted this system after five years of study.

stereophonic sound, an additional monophonic sound programme (second audio programme or "SAP") service, and a narrowband or data channel for station operation purposes. At this time, the BTSC system as described in section 6 is preferred for use in the United States but improved systems are possible.

## 2. System requirements

In addition to the capacity requirements of the additional sound or information channels, the following requirements should be fulfilled:

- the system should be compatible with existing receivers and networks;
- the system should not involve any increase in the bandwidth of a television channel;
- the additional sound or information channel should have at least the same coverage area as that of the picture channel;
- the system for an additional sound channel should provide high-quality reproduction of both channels and stereophony;
- for receiving the additional sound channel, an ordinary receiver and a relatively simple adapter can be used.

## 3. Two-sound carrier FM system

The system is recommended for analogue multisound in PAL television systems B, G and H. Its principal specifications are given in the proposal for a new Recommendation shown in Annex I.

### 3.1 Principles of the system

The two-carrier FM system provides an additional sound carrier for the second sound signal. The additional sound carrier is located at a higher frequency than the normal sound carrier and is set at a level lower than that of the first sound carrier.

### 3.2 Transmitter and receiver aspects

The system is implemented using simple well established techniques. Receiver cost is low. The circuits for the two sound carriers are identical for transmitters as well as receivers.

### 3.3 Stereophonic coding

The first sound channel is modulated by signal M, equal to one half of the sum of the left-hand signal A and the right-hand signal B and the second channel by signal B only. This coding is chosen in order to improve the audio S/N ratio in receivers which use intercarrier demodulation.

### 3.4 Coverage area

The service area of both sound channels is larger than the service area of the vision signal. Even when propagation conditions are unfavourable (long-distance and multipath propagation), reception is still satisfactory for both sound channels.

3.5 Quality of the two sound channels

The following figures are typical for modern television receivers using inter-carrier demodulation [CCIR, 1986-90b]. Further improvements are possible, e.g. with split-sound receiver types, synchronous detection or audio-frequency companding:

- weighted signal-to-noise ratio (CCIR Recommendation 468 peak-to-peak): 50 dB for normal pictures, 45-50 dB for those pictures causing the most interference (the possibilities for the use of an additional compatible companding system to improve the S/N ratio are being studied);

- cross-talk attenuation: dual-sound mode >62 dB;

- cross-talk attenuation: stereophonic mode >36 dB;

- audio-frequency response: $< \pm$ 0.7 dB between 40 Hz to 15 kHz;

- harmonic distortion: <1%.

3.6 Compatibility

More than 10 million dual sound receivers are in use in 1989 in the Federal Republic of Germany. The system is operating without compatibility problems in the whole receiver population (i.e. including receivers not designed for the dual sound service) [CCIR, 1986-90b]. A similar situation exists in other countries which have introduced the system.

3.7 Data transmission

Laboratory tests carried out in Italy [CCIR, 1986-90c] have shown that during monophonic television programmes the second sound channel is available to transmit data at 4.8 kbit/s without affecting the picture or sound. In this particular service the data signal is sent via the peritelevision connector to a home computer [CCIR 1986-90d, e], [d'Amato et al., 1987].

4. NICAM 728 digital carrier system

This system is recommended for digital multisound in PAL television systems B, G, H and I, see Recommendation 707. A summary specification is given in Annex II to that Recommendation.

4.1 Principal characteristics

The NICAM 728 system uses an additional sound carrier located at a higher frequency than the normal sound carrier and set at a level lower than that of the first carrier. This second carrier is modulated using a process of quadrature phase shift keying by a digital signal with a bit rate of 728 kbit/s. This signal can carry either two high quality sound signals, which may be a stereo pair or independent mono, or one such sound signal with data, or data alone; associated signalling bits indicate which option is in use. The use of these signalling bits is not defined to allow for the possibility of other sound/data combinations in the future. There is also a provision for 11 kbit/s of ancillary data. High quality sound is linearly encoded at 32 kHz sampling rate and 14 bits per sample, the latter being reduced to 10 bits per sample by a process of near-instantaneous companding as defined in Report 647.

The system has the following features:

- it provides two high-quality sound channels in addition to the existing analogue sound channel;
- it is sufficiently rugged to ensure that reception of the vision signal fails before reception of the digital sound signal under difficult reception conditions;
- it gives satisfactory compatibility with existing services and receivers in over-the-air transmission and in distribution on cable systems;
- the sound coding is identical with that of one option available in the MAC/packet family of systems [CCIR, 1988].

4.2 Coverage area

Field trials with systems B, G and I show that the digital signal can be transmitted from a main station and travel acceptably through up to seven television transposers. In addition, the digital signal was found to be rugged against impaired reception due to weak signal conditions, multipath propagation and ignition interference [CCIR, 1986-1990f, g, h]. The service area for satisfactory reception is therefore equal to or greater than the area for satisfactory colour television reception.

4.3 Quality of the digital channels

The signal-to-weighted-noise ratio of a digital sound system is limited only by the inherent quantization accuracy of the digital sound coding system used. The main characteristics of the 14/10 near-instantaneous companding system are given in section 4.1.

The cross-talk attenuation between the two additional digitally coded sound channels is found to exceed 70 dB. Cross-talk attenuation from the normal mono sound channel into the digital sound channel also exceeds 70 dB. Such cross-talk is due only to implementation imperfections in the analogue parts of the transmission chain.

In the digital system, harmonic distortion is not inherent. Any harmonic distortion that is produced will depend upon the performance of the associated analogue circuits, including the A/D- and D/A- converters.

4.4 Compatibility with existing receivers

The introduction of the digital sound carrier could in principle cause interference into either the analogue sound or the picture. Over-air tests with system I and systems B and G have been carried out [CCIR, 1986-90i, j]. No significant problems with patterning on the picture were reported. Test results are similar to those obtained with the analogue two-carrier system.

Regarding interference from the digital signal into the analogue sound channel, results from over-air tests and supplementary laboratory measurements show that no serious problems are envisaged [CCIR, 1986-90f].

Results from operation of the digital system show no significant problems with existing receivers [CCIR, 1986-90k, l].

## 4.5 Compatibility with existing networks

Relating to interference from the NICAM 728 signal on a lower adjacent channel into the picture signal of a wanted channel, laboratory tests, tests in cable networks, off-air tests as well as regular operation with system I, B and G show no significant problems [CCIR, 1986-90f, k, m, n, o, h, p, q].

For the case of interference from the picture signal of an upper adjacent channel interfering into the NICAM channel of the wanted signal, tests in the United Kingdom as well as regular operation in Denmark demonstrated no significant problems with upper adjacent channel interference [CCIR, 1986-90k, p, q]. Any such interference could be limited by a slight modification of the vestigial sideband filter for the upper adjacent channel [CCIR, 1986-90r].

Calculations and laboratory measurements in Finland relating to interference from system B/PAL with NICAM 728 into system D/SECAM has demonstrated that the protection ratio to be applied in planning decreases 2 dB since the level of the FM sound carrier is lowered from -10 to -13 dB relative to the peak vision carrier [CCIR, 1986-90s].

Calculations and laboratory tests [CCIR, 1986-90f, t] have shown that the use of the digital sound system in system I will comply with the protection ratios given in CCIR Recommendation 655 for continuous and tropospheric interference for this system. Protection ratios applicable to co-channel interference from system I with NICAM 728 into system L are under study in the United Kingdom, in Study Group 11 and by the EBU.

Report 1214 gives information on the protection ratios applicable to situations with a digital signal involved.

## 4.6 Receiver population

Within one year of the introduction of the digital system in 1988 in Denmark, Finland and Sweden, more than 200,000 receivers for this system were in use in these countries.

## 4.7 Transmitters

The system can be easily implemented in both new and older transmitters. Experience in Denmark shows that in order to allow for margins in large cable networks, the power spectrum of the NICAM 728 signal at the output of the terrestrial transmitter should be within ±1 dB relative to the ideal response for frequencies in the range +5.85 MHz ±150 kHz relative to the vision carrier [CCIR, 1986-90k].

# 5. FM-FM system for NTSC television system M

## 5.1 Principles of the system

Specifications of the first sound channel and the vision channel in the system are the same as that of television system M.

The FM-FM system makes use of a sub-carrier in the normal sound channel. The frequency of the sub-carrier, which is frequency-modulated by the second sound signal, is chosen to be the second harmonic of the picture line frequency, because this reduces interference from the picture into the second sound channel.

The second sound channel has an audio-frequency response of up to 14 kHz. The system can be used for high quality transmission of either two independent programmes or one stereophonic programme.

The system specifications are given in Appendix I to Annex I to this report.

5.2 System performance

5.2.1 Receivers

Laboratory and broadcasting tests made in Japan showed that, with suitable design of the sound intermediate-frequency circuits, inter-carrier receivers were suitable for reception of a second sound channel having an audio-frequency response of up to 14 kHz.

5.2.2 Stereophony

In the FM-FM system, compatibility with existing standards necessitates modulation of the first and the second channels by the signals M and S respectively, as defined in Recommendation 450.

5.2.3 Coverage area

In the FM-FM system, the first sound channel has the same coverage area as the ordinary monophonic transmission and the second sound channel has the same coverage area as that of the picture channel.

Even when propagation conditions are unfavourable (long-distance multipath propagation), the system is still satisfactory for stereophony.

The FM-FM system was examined with respect to the effects on the audio-frequency signal-to-noise ratio under the conditions of line and precision-offset [CCIR, 1978-82].

5.2.4 Quality of the second channel

Signal-to-weighted noise ratio, including buzz:

- 51 dB (inter-carrier receiver).

Cross-talk from the main sound channel into the second sound channel:

- Under normal operation, the cross-talk attenuation from the first sound channel into the second sound channel was found to exceed 55 dB.

Harmonic distortion:

- 44 dB (inter-carrier receiver).

5.2.5 Stereophonic quality

The cross-talk attenuation between the stereophonic signals A and B is better than 30 dB (0.4 to 7 kHz; inter-carrier receiver).

5.2.6 Compatibility (vision and sound)

5.2.6.1 Compatibility with existing receivers

In the operational service in Japan, no complaint concerning interference into the picture channel has been reported and very few complaints of interference into the first sound channel were received. Since it is estimated that more than thirty million viewers watch the programmes every day, the compatibility of the FM-FM system is said to be good.

In the case of the digital broadcast facsimile system described in § 5.3.1, there is no interference to the picture concerned as well as to both sound channels of existing receivers of FM-FM multi-channel sound systems, if the maximum frequency deviation of the main sound carrier by the facsimile sub-carrier is limited to ±2 kHz.

In the case of the analogue broadcast facsimile system described in § 5.3.2, the maximum frequency deviation of the main sound carrier by the facsimile sub-carrier has to be limited to ±10 kHz. With this deviation interference occured in very few receivers, but if a trap for the sixth line frequency multiple is inserted into the receiver, this problem would be eliminated.

5.2.6.2 Compatibility with existing networks

In Japan, with rebroadcasting by 1355 stations operated using the FM-FM system, no interference to the picture and the first sound channel was detected even when five rebroadcasting stations were operated in cascade. Interference from the picture signal into the second sound channel did not increase significantly with inter-carrier reception.

In broadcasting the FM-FM system operating in Japan no increase in interference to the picture and the first sound channel from adjacent channels and co-channels was observed.

5.3 Broadcast facsimile systems

Broadcast facsimile signals can be transmitted by a DQPSK- or FM-modulated sub-carrier to the main sound carrier simultaneously with the sound signal carried by the sound sub-carrier.

5.3.1 Digital system

A digital broadcast facsimile system developed in Japan [CCIR, 1986-90u] emphasizes compatibility with G3 (conforming to CCITT Recommendation T.4) apparatus and provides for facsimile transmission of ISO A4 page size text at 9600 bit/s net data signalling rates.

As to the signal format, the system employs a packet type signal which incorporates a majority logic decodable (272, 190) shortened difference set cyclic code for error correction.

The transmission characteristics are shown in Appendix I to Annex I of this report.

5.3.2 Analogue system

An analogue broadcast facsimile system has been developed in Japan [CCIR, 1986-90u]. It has a 2.5 minutes constant transmission time per page of ISO A4 size. Transmission of colour documents takes the same time as transmission of monochrome documents. In the colour transmission mode, chrominance components compressed to a 1/2 time axis are transmitted every two lines by substituting for the even numbered luminance lines. By this substitution a resolution corresponding to the normal mode on G3 apparatus is obtained. In the monochrome mode, a resolution corresponding to the fine mode on G3 apparatus is obtained.

The transmission characteristics are shown in Appendix I to Annex I of this report.

## 6. BTSC FM-AM system

### 6.1 Principles of the system

The BTSC system for multichannel television sound consists of an M signal derived from the sum of the left and right sound signals. This signal is compatible with monophonic receivers in that it has the same preemphasis and peak sound carrier deviation as a monophonic signal. An S signal is derived from the difference of the left and right sound signals and is compressed to make an S' signal that modulates a suppressed-carrier amplitude modulated sub-carrier. The frequency of the sub-carrier is 31 468 Hz, the second harmonic of a pilot-tone of line frequency. The pilot-tone is used to synchronize the encoding of the S sub-channel.

The compressor forms the encode portion of a compressor/expander (compander) used to reduce the noise level on the S sub-channel to an acceptable level. This sub-channel noise level is higher in television stereophonic operation than in FM sound broadcasting because virtually all NTSC system M television receivers employ intercarrier sound detectors to recover the sound signal.

To maintain maximum stereophonic separation, the S sub-channel encoder (compressor) must be matched in amplitude exactly by the corresponding decoder (expander) in the receiver. In order to maintain the required amplitude match, each portion of this compandor system (both the encoder and decoder) must have a common amplitude reference. The absolute deviation of the transmitter is this reference.

The system specifications are given in Appendix II of Annex I to this report.

### 6.2 Additional signals

Provision is also made in the BTSC for the transmission of additional signals on the sound carrier.

One signal is a monophonic sound sub-channel (separate audio programme: SAP) at five times the horizontal line frequency. It uses noise reduction encoding similar to what is used on the S channel, and provides a 50 Hz to 10 kHz audio frequency response. It is intended for reception by a wide audience on stereophonic receivers.

Another signal is a non-programme sound and data channel at 6.5 times the horizontal line frequency. It is a voice bandwidth sub-channel intended for operational (non-public) use by the broadcast authority.

### 6.3 System performance

The BTSC system was chosen by the United States as its preferred system following comprehensive evaluation [EIA/BTSC, 1983].

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ANNEX I

PROPOSAL FOR A DRAFT NEW RECOMMENDATION

TRANSMISSION OF MULTISOUND IN TERRESTRIAL TELEVISION SYSTEM M

(Question 47/10, Study Programme 47A/10)

The CCIR,

CONSIDERING

(a) the increasing requirement world-wide for suitable means of broadcasting stereophonic and/or multi-channel sound and/or data from terrestrial television transmitters;

(b) the technological development in this area and in particular the relative merits of various possible analogue and digital methods, and described in Report 795;

(c) the fact that the United States and Canada have used the BTSC FM-AM system as their preferred system beginning in March 1984 and that a public service of bilingual and stereophonic television programmes using the FM-FM system has been operation in Japan since 1978;

(d) the advantage of low-cost analogue circuitry for multisound television receivers for system M;

(e) the desirability of establishing standards for the introduction of stereophonic and/or multi-channel should for the terrestrial television broadcast services using system M,

RECOMMENDS

1. that if analogue multisound is to be introduced in terrestrial broadcasting in countries using NTSC television system M, it should be implemented using the FM-FM or BTSC system, and be implemented as defined in Appendices I and II to this draft new Recommendation.

Note 1 - The transmission systems described, in some cases, can be used for data services. Where applicable, reference to these data services will be found in the annexes containing the system specifications.

Note 2 - A modified two-sound carrier FM system similar to the system described in Table I of Recommendation 707, Annex I, has been introduced in the Republic of Korea for system M.

## Appendix I to Annex I

## System specifications for the FM-FM system

TABLE I

Transmission characteristics of the FM-FM system
(Television system M)

| | |
|---|---|
| 1. *The first channel* (the same as the monophonic channel) | |
| Maximum frequency deviation of the main carrier (kHz) | ± 25 |
| Audio-frequency range (Hz) | 50 to 15 000 |
| Pre-emphasis ($\mu$s) | 75 |
| 2. *The second channel* | |
| Sub-carrier frequency | second harmonic of the line frequency |
| Maximum frequency deviation of the sub-carrier (kHz) | ± 10 |
| Frequency deviation of the main carrier by the sub-carrier (sub-carrier level) (kHz) | ± 15 |
| Audio-frequency range (Hz) | 50 to 14 000 |
| Pre-emphasis ($\mu$s) | 75 |
| 3. *Stereophony* | |
| ($A$ + $B$) channel | the same as the first channel |
| ($A$ − $B$) channel | the same as the second channel except the sub-carrier level |
| Frequency deviation of the main carrier by the sub-carrier (sub-carrier level) (kHz) | ± 20 |
| The left-hand signal produces a deviation in the same direction in both the sub-carrier and the main carrier | |
| Compensation for receiver delay time ($\mu$s) | 20 |
| 4. *Control signal* | |
| Sub-carrier frequency | 3.5 times the line frequency |
| Modulation frequency: | two sound programmes: 922.5 Hz<br>stereophonic programme: 982.5 Hz |
| Modulation | 60% (AM) |
| Maximum frequency deviation of the main carrier by the control signal sub-carrier (kHz) | ± 2 |

TABLE II

Transmission characteristics of digital facsimile broadcast multiplexed with the FM-FM system

| | |
|---|---|
| Facsimile channel | |
| Sub-carrier frequency | 70.804 kHz |
| Bandwidth | 12.8 kHz |
| Modulation method | DQPSK |
| Data signalling rates | 16 kbit/s |
| Spectrum shaping | 60% square root cosine shaping (equally shared between transmitter and receiver) |
| Data scrambling | 11th M-sequence, pseudo-random noise signal $g(X)=X^{11}+X^{2}+1$ |
| Maximum frequency deviation of the main carrier | ± 2kHz |

TABLE III

Transmission characteristics of analogue facsimile broadcast multiplexed with the FM-FM system

| | |
|---|---|
| Facsimile channel | |
| Sub-carrier frequency | 94.4 kHz |
| Bandwidth | 63 kHz |
| Modulation method | FM |
| Frequency deviation | +16kHz, -8 kHz |
| Pre-emphasis | 50µs |
| Maximum frequency deviation of the main carrier | ±10 kHz |

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## APPENDIX II TO ANNEX I

## SPECIFICATIONS FOR THE BTSC FM-AM SYSTEM

1.1 A main (sum) channel containing audio frequencies from 50 to 15 000 Hz, with peak modulation corresponding to +/- 25 kHz deviation and with 75 microsecond pre-emphasis.

1.2 A pilot sub-carrier at the horizontal line frequency 15 734 Hz +/-2 Hz, peak deviation of +/- 5 kHz. This pilot sub-carrier is to be frequency locked to the horizontal line frequency of the transmitted video signal. Stations not transmitting multi-channel sound according to this specification must limit modulation of the main sound carrier at 15 734 Hz +/- 20 Hz, to no more than 0.125 kHz deviation to prevent false activation of stereo mode reception in receivers.

1.3 A stereophonic sub-carrier at the second harmonic of the pilot sub-carrier, double sideband amplitude modulated, suppressed carrier, with the stereophonic difference signals from 50 to 15 000 Hz. This sub-carrier must cross the time axis with a positive slope simultaneously with each crossing of the time axis by the pilot carrier, and within 3 degrees (approximately +/- 0.53 $\mu$s) of the zero crossings of the pilot sub-carrier. During the time period in which a left-only signal applies, the main channel modulation must cause an upward deviation of the sound carrier. The unmodulated sub-carrier must be suppressed to a level less than 0.25 kHz deviation of the sound carrier.

1.4 A frequency-modulated second audio programme sub-carrier, frequency locked to the fifth harmonic of the horizontal line frequency (pilot sub-carrier). When modulated, the centre frequency should nominally be that of the the fifth harmonic of horizontal line frequency with a tolerance of +/- 500 Hz. The frequency response of this channel is 50 to 10 000 Hz and modulates the sub-carrier with a peak deviation of +/- 10 kHz. This sub-carrier modulates the sound carrier with a peak deviation of +/- 15 kHz.

1.5 Audio amplitude and spectral compression (for complementary expansion in receivers) providing approximately 30 dB of background noise improvement in both the stereophonic sub-channel and the second audio programme channel. The compression has the following characteristics, where f is expressed in kHz (see also Figure 1).

1.5.1 Fixed pre-emphasis (F(f)) whose transfer functions is as follows:

$$F(f) = \frac{(jf/0.408)+1}{(jf/5.230)+1} \cdot \frac{(jf/2.19)+1}{(jf/62.5)+1}$$

1.5.2 Wideband amplitude compression wherein:

(a) The decibel gain (or loss) applied to the audio signal during encoding is equal to minus one times the decibel ERMS value of the encoded signal (the result of the encoding process), weighted by a transfer function (P(f)) as follows:

$$P(f) = \frac{(jf/0.3254)}{((jf/0.0354)+1) \cdot ((jf/2.09)+1)}$$

where ERMS (Exponentially time-weighted root mean square value equals

$$\text{ERMS value} = \sqrt{\frac{1}{T}\int_{-\infty}^{t} S^2(u)e^{-(t-u)/T}\,du}$$

and where S(u) is the waveform in question, a function of time, T is the exponential time-weighting period, and t is the time at which the ERMS value is computed.

(b) The exponential time weighting period $T_1$ of the ERMS detector referred to above in (a) is 34.7 ms.

(c) The zero decibel reference ERMS value for the encoded signal referred to above in (a) is 8.99% modulation of the sub-carrier at 0.300 kHz.

1.5.3 Spectral compression wherein:

(a) The transfer function S(f,b)) applied to the audio signal during encoding is:

$$S(f,b) = \frac{1 + (jf/F)\cdot(b+51)/(b+1)}{1 + (jf/F)\cdot(1+51b)/(b+1)}$$

where $b=10^{D/20}$
F=20.1 kHz
D = decibel r.m.s. value and b is the decibel ERMS value of the encoded signal (the result of the encoding process) weighted according to a frequency transfer function Q(f) as follows:

$$Q(f)=\frac{(jf/5.86)^3}{((jf/7.66)^2+(jf/7.31)+1) \cdot ((jf/26.9)+1)\ (jf/3.92)+1)}$$

(b) The exponential time weighted period $T_2$ of the ERMS detector referred to above in 1.5.3(a) is 11.4 ms.

(c) The ERMS zero decibel reference for the encoded signal referred to above in 1.5.3(a) is 5.16% modulation of the sub-carrier at 8 kHz.

1.5.4 Overmodulation protection functionally follows the functions of 1.5.1, 1.5.2, and 1.5.3 above.

1.5.5 Band limiting to appropriately restrict bandwidth functionally follows the functions of 1.5.1, 1.5.2, and 1.5.3 above.

1.6 A non-programme related sound sub-carrier at 6.5 times the horizontal line frequency (pilot sub-carrier frequency) with a tolerance of +/- 500 Hz. The peak deviation of this sub-carrier is limited to +/- 3 kHz. This sub-carrier may be used to transmit audio or telemetry and control data signals. When the second audio programme is not transmitted a sub-carrier frequency between 47 and 120 kHz may be used for which the deviations are not specified.

1.7 A limit of 73 kHz instantaneous deviation of the aural carrier by the sum of all multi-channel sound signals.

## BIBLIOGRAPHY

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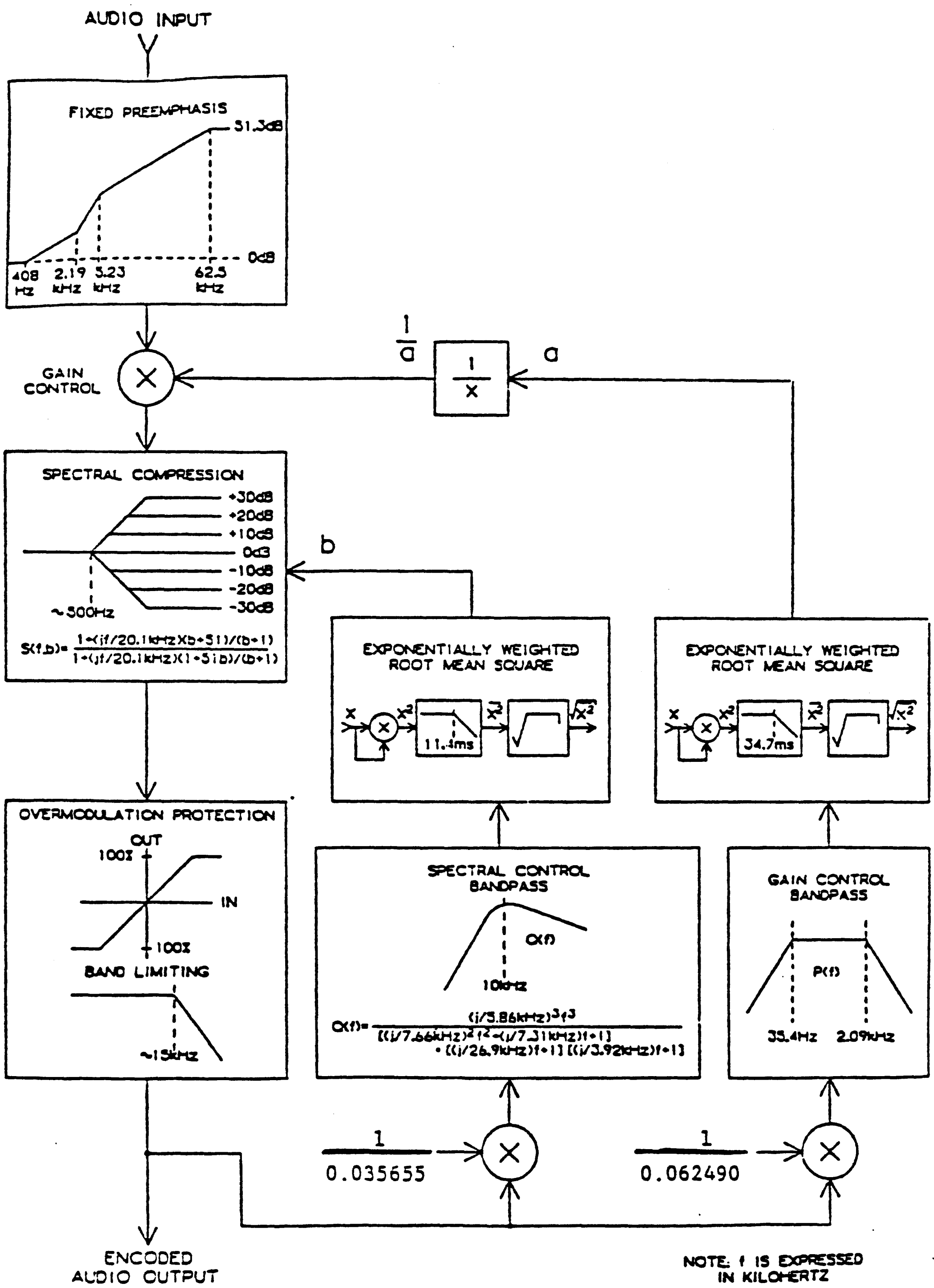


Note: Certain values in this figure are approximate. Exact values are found in the accompanying text.

**FIGURE 1**

STEREOPHONIC DIFFERENCE AND SECOND PROGRAM AUDIO SIGNAL ENCODING