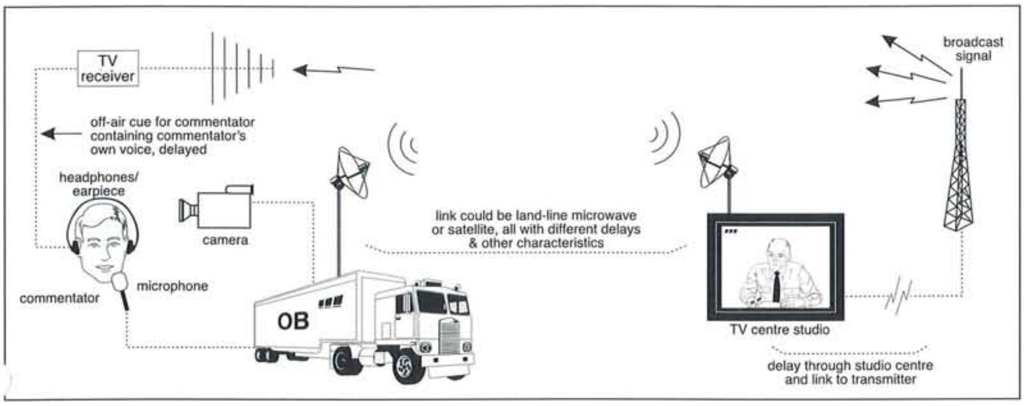
DIGITAL AUDIO

PRODUCT SUMMARY

Bel 7410 Mix-Minus Synchroniser

The Bel 7410 has been designed primarily to provide a clean feed to commentators at outside broadcasts or remote locations who monitor their own broadcasts from the off-air programme. However, the main problem is that commentators hear their own voice in the off-air feed, subject to the delay incurred by passing through the broadcast chain. Quite often the delay may only be 20mS or so - noticeable, but not very disturbing. Satellite links introduce a further 250mS delay and low bit-rate coding and decoding (used with Digital Audio Broadcasting) can introduce delays around 300mS. These will have a highly distracting effect even on an experienced commentator.



Removing the commentary from the mix could, theoretically, be done by inverting the commentary and adding it to the mix so that it cancels out. Unfortunately, the commentary together with the programme is likely to have been modified since it was created at the OB site. It will have been delayed, and may have been subject to changes in level and equalisation.

Therefore to get normal mix-minus cancellation, the commentary at the OB site has to be modified in exactly the same way as the "off-air" feed. The 7410 does this, adjusting delay, gain and equalisation completely automatically, presenting a mirror image of the "off-air" signal thereby producing a "mix-minus" feed for the commentator.

The 7410 adapts automatically to changing circumstances and saves both expensive set-up time and the cost of cue/communication lines between the remote site and the studio.

SPECIFICATIONS - MODEL 7410			
General		Inputs & Outputs	
Maximum delay	1 Sec	Input 1 (off-air)	Stereo analog
Frequency response	100Hz - 10kHz max	Input 2 (commentary feed)	Mono analog
Input dynamic range	100dB	Output (mix-minus)	Mono analog
Signal to noise ratio	90dB RMS 100Hz - 10kHz	Inputs	Electronically balanced 25kΩ
Distortion	less than 0.03% at 1kHz	Outputs	Electronically balanced 50Ω Max. drive capability +18dBu
Conversion accuracy	A/D 18 bit Delta Sigma 64 x oversampled D/A 20 bit 8 x oversampled	Power Supply	
		Supply voltages	110 - 120 VAC, 220 - 240 VAC 50/60Hz
Sampling rate	24kHz internal reference		
Commentary attenuation	Greater than 25dB		
Commentary reduction attack time	Approx. 1 Second	Power consumption	50W
		Dimensions	482w x 44h x 345d
Remote Interface (Mute/Man)	GPI	Net Weight	7kg

Technical specifications subject to change without notice.

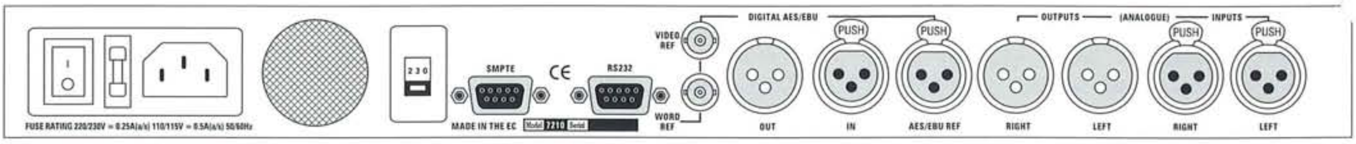
Bel 7000 Series Audio Delay Synchronisers

Bel 7110

The cost effective Bel 7110 has analog inputs/outputs and provides 0-660msecs (1-15 frames) of audio delay, in stereo, mono or dual mono. Eight user-defined settings may be stored for future use and a lock function prevents inadvertent operation. The LCD indicates the mode selected together with delay times which may be incremented in samples, msecs, fields or frames. The field and frame step values may be changed to reflect PAL or NTSC operation. Bypass is automatically engaged if a power failure occurs.

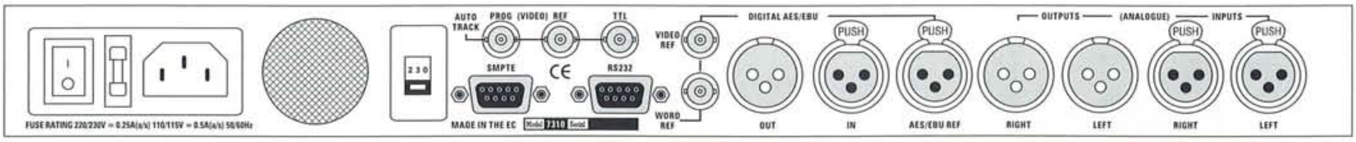
Bel 7210

Features of the Bel 7210 are similar to the Bel 7110 with additional memory (1.3 secs), SMPTE delay, an AES/EBU I/O with nominal sampling frequencies of 32-48kHz and clock referencing to AES, Word or Video. There is a "soft-nudge" facility for glitch free adjustments in real time. An RS232 serial interface is provided for optional remote control via a standard PC with Windows 95 software. A secondary application is for sample rate conversion, A/D and D/A conversion.



Bel 7310

As well as incorporating the features of both models 7110 and 7210, the Bel 7310 provides three modes of automatic delay adjustment in response to external signals: [1] source & reference video sync, [2] an active low TTL pulse, [3] RS232 commands from a PC. All modes operate silently in real-time with a choice of algorithms, enabling differing audio and vision transmission paths to be synchronised without manual adjustment. Alternatively, a glitch free manual delay may be set in conjunction with the auto-track mode.



SPECIFICATION	MODEL 7110	MODEL 7210	MODEL 7310
Delay	0-660 msec	0-1.3 sec	0-1.3 sec
Delay increments	1 sample, 1 msec, 20 msec, 40 msec, 1 field, 1 frame	sample, msec, field, frame	sample, msec, field, frame
Video system	PAL or NTSC	PAL or NTSC	PAL or NTSC
Frequency response	20Hz - 20kHz ± 1dB	20Hz - 20kHz ± 1dB	20Hz - 20kHz ± 1dB
Dynamic range	96dB	108dB	108dB
Signal to noise ratio	-90dB rms 20Hz - 20kHz	-100dB rms 20Hz - 20kHz	-100dB rms 20Hz - 20kHz
Distortion	less than 0.015% at 1kHz	less than 0.015% at 1kHz	less than 0.015% at 1kHz
Conversion accuracy	A/D 16bit Linear Sigma Delta 64 x oversampled D/A 16 bit 8 x oversampled	A/D 18/20bit Linear Sigma Delta 64 x oversampled D/A 20 bit 8 x oversampled	A/D 18/20bit Linear Sigma Delta 64 x oversampled D/A 20 bit 8 x oversampled
Sampling rate	48kHz	48kHz	48kHz
Inputs	Electronically balanced 25kΩ	Electronically balanced 25kΩ	Electronically balanced 25kΩ
Outputs	Electronically balanced 600kΩ Max. drive capability +18dBm	Electronically balanced 600kΩ Max. drive capability +22dBm	Electronically balanced 600kΩ Max. drive capability +22dBm
Auto-track			Mode 1 (Video sync) 39.5msec Mode 2 (TTL) 1.32secs
Digital input		24 bit AES/EBU 32kHz - 48kHz	24 bit AES/EBU 32kHz - 48kHz
Digital output		24 bit AES/EBU 48kHz	24 bit AES/EBU 48kHz
Reference		1. Internal clock 2. External word clock (48kHz) 3. External AES/EBU (48kHz) 4. Ext video (1Vpk) Auto PAL/NTSC	1. Internal clock 2. External word clock (48kHz) 3. External AES/EBU (48kHz) 4. Ext video (1Vpk) Auto PAL/NTSC
RS232 serial ports		9.6kbaud, 8 data bits, 1 stop bit no parity. BEL 7210 serial protocol.	9.6kbaud, 8 data bits, 1 stop bit no parity. BEL 7310 serial protocol.
SMPTE in/out		Audio delay follow	Audio delay follow
Power requirements	115/230VAC 50/60Hz	115/230VAC 50/60Hz	115/230VAC 50/60Hz
Power consumption	30W	35W	35W
Dimensions	482w x 44h x 240d	482w x 44h x 345d	482w x 44h x 345d
Net weight	6kg	7kg	7kg

Technical specifications subject to change without notice.

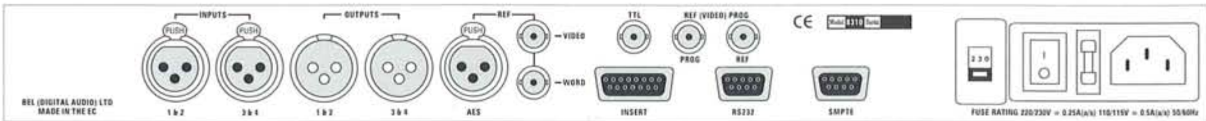
Bel 8110 Eight Channel Analog Audio Delay Synchroniser

The cost effective Bel 8110 audio delay is designed specifically for synchronising audio to video in post production areas where video delays result from VT editing and video effects units etc. It is intended for use where the video delay is known and is likely to be constant or when frequent changes of delay are not required. The Bel 8110 may be used in a fixed location with each channel dedicated to a specific item of video equipment. Independent A/D and D/A converters are employed for each channel. Each stereo pair of delay lines is switchable in fields (PAL / NTSC) 0, 1, 2, 3, 4, 5, 6, 7, 8 & 9 (4.5 frames). An input level pre-set potentiometer, a signal-present and peak LED are fitted for each stereo pair. The Bel 8110 user controls are recessed to prevent inadvertent operation. Master signal bypass is effected by relays in the event of power loss.

SPECIFICATIONS - MODEL 8110	
Frequency response	20Hz - 20kHz ± 1db
Dynamic range	100dB
Distortion	less than 0.015% at +8dBm/1kHz
Audio inputs	Electronically balanced
Audio input level	+6dBm nominal
Audio input impedance	25 kΩ
Conversion	20 bit Sigma Delta 64 x oversampling
Sampling frequency	48kHz
Audio outputs	Electronically balanced
Audio output level	+6dBm nominal, +18dBm max
Audio output impedance	50 Ω
Power requirements	115/230VAC 50/60Hz
Power consumption	35W
Dimensions	482w x 44h x 200d
Net weight	5kg

Technical specifications subject to change without notice.

Bel 8310 AES/EBU Audio Delay Synchroniser



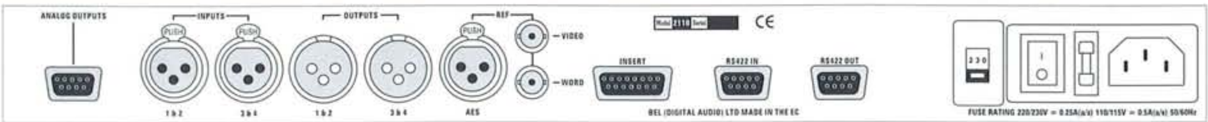
The Bel 8310 has 2 AES/EBU inputs and provides 1.3 secs (1-30 frames) of audio delay, in 2 stereo pairs or 4 mono channels. Parameters of the stereo pairs may be set individually whilst mono channels are set as a block. Eight user-defined settings may be stored, a lock function is also provided to prevent inadvertent operation. The LCD indicates the mode selected together with delay times which may be incremented in samples, msecs, fields or frames. The field and frame step values may be changed to reflect PAL or NTSC operation. Bypass is automatically engaged in the event of power failure. Input sampling frequencies are from 32-48kHz. Clock referencing to AES, Word and Video are provided as well as an RS232 serial remote interface for external control via a PC. A SMPTE delay is also incorporated.

The Bel 8310 also provides three modes of automatic delay adjustment in response to external signals: [1] source & reference video syncs, [2] TTL pulse, [3] RS232 commands from a PC. All modes operate silently in real-time with a choice of algorithms enabling changing audio and video transmission paths to be synchronised without manual adjustment. Alternatively a glitch free manual delay may be set in conjunction with the auto-track mode.

SPECIFICATIONS - MODEL 8310	
Delay	0-1.3 secs
Delay increments	sample, msec, field, frame
Video system	PAL or NTSC
Input	24 bit AES/EBU 32kHz-48kHz
Output	24 bit AES/EBU 48kHz
Auto-track	Mode 1 (Video sync) 39.5 msec Mode 2 (TTL) 1.32 secs
Reference	1. Internal clock 2. External word clock (48kHz) 3. External AES/EBU (48kHz) 4. External video (1V pk) auto PAL/NTSC
RS232 serial port	9.6kbaud, 8 data bits, 1 stop bit no parity. BEL 8310 serial protocol
SMPTE in/out	Audio delay follow
Power requirements	115/230VAC 50/60Hz
Power consumption	35W
Dimensions	482w x 44h x 200d
Net weight	7kg

Technical specifications subject to change without notice.

Bel 2110 AES/EBU Audio 'Shuffler'



The main function of the Bel 2110 is to offer a cost effective solution to modifying AES/EBU audio signals in the digital domain. The main areas of use are where occasional or minor corrections to standard practice are required such as in transmission, racks and VT areas. The Shuffler is designed to allow 4 AES/EBU audio input signals to be selected, modified and then routed to 4 digital outputs. Four high quality analog outputs are provided which reflect the digital outputs. A headphone monitor jack is also provided on the front panel.

The unit can be controlled by front panel switches or remotely. Two modes of operation are possible:

In the normal mode each input can be inverted and its level adjusted in 0.5dB steps to ±10dB. Only one input can be routed to one output. In the mix mode each input can be inverted and its level adjusted but any combination of the inputs can be routed to each output. All setup parameters may be stored in the eight user memories and a safety lock avoids inadvertent operation of the front panel controls.

Standard external referencing is AES/EBU 48kHz with video/word as an optional extra. An optional communication port may be incorporated which will allow signals to be routed to and from the Shuffler from external equipment such as equalisers, compressors or delay units to inputs or outputs. Full remote operation of the Bel 2110 is possible via an RS422 port on a PC with optional Windows 95 software.

SPECIFICATIONS - MODEL 2110	
Digital	
Inputs	2 pairs stereo AES/EBU
Audio sampling frequency	32-50kHz, nominally 48kHz
Audio outputs	2 pairs stereo AES/EBU
Word length	Maximum 24 bit
Reference	48kHz AES (video/word clock optional)
Auxiliary I/O	External equipment bus (optional)
Analog	
Analog audio outputs	4 differential outputs
D/A	20 bit (better than 100dB dynamic range
Level	+6dB out for -9dB digital in (max 15dB adj)
Output impedance	50Ω
Monitor output	2 channel analog with source selector Output impedance 8Ω min
General	
Remote interface	RS422, 9.6kBaud
Power requirements	115/230VAC 50/60Hz
Power consumption	35W
Dimensions	482w x 44h x 200d
Net weight	7kg

Technical specifications subject to change without notice.

Bel 5110 Stereo Broadcast Profanity Delay

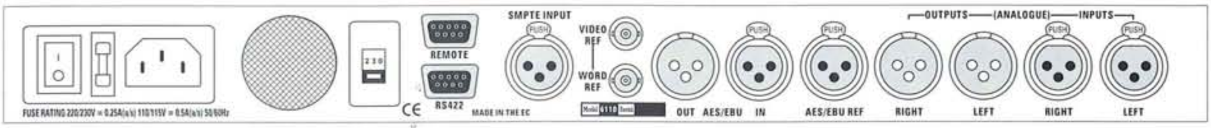
The Bel 5110 provides 7 Secs of stereo audio delay which is the optimum time required to identify and remove undesirable program content during 'phone-ins or live studio productions. Utilising advanced technology the "auto-catch-up" facility enables delay to be introduced in real time, allowing the program to flow without interruption. Two algorithms are available for speech or music, switchable from the wired remote control (optional). This also contains the "dump" and "delay in/out" switches. When the "dump" button is pressed the Bel 5110 will provide automatic cart insert via the cart player interface also returning the program output to real-time. A fail-safe bypass system is also incorporated in the event of power failure. All controls are via GPI 15 pin socket on rear of unit or optional remote control. Analog or AES/EBU I/O cards are available and should be specified when ordering.

Optional Remote Control

SPECIFICATIONS - MODEL 5110	
Analog	
Audio inputs	Electronically balanced
Audio input level	Internal headroom select
Audio input impedance	25kΩ
Audio outputs	Electronically balanced
Audio output level	Internal headroom select
Audio output impedance	50Ω
Conversion	16 bit Sigma Delta
Frequency response	20Hz - 20kHz ±1dB
Dynamic range	96dB
Distortion	less than 0.015% at 8dBm/1kHz
AES/EBU	
Sampling frequency	48kHz
Data format	16 bit
External reference	AES/EBU 48kHz
General	
Full delay	7 Secs
Cart interface	Contact open/closure
Power requirements	110/220VAC 50/60Hz
Power consumption	35W
Dimensions	482w x 44h x 240d
Net weight	6kg

Technical specifications subject to change without notice.

Bel 6110 Audio Lay-off Recorder



The Bel 6110 lay-off recorder operates in the timecode "follow" mode or via an RS422 serial interface, in PAL or NTSC formats. In the "follow" mode it records and replays audio in mono or stereo, in synchronisation with other equipment when fed with external SMPTE timecode. Effectively, it provides two additional tracks of audio in order to simplify editing when audio needs to be cross faded and not edited with the video.

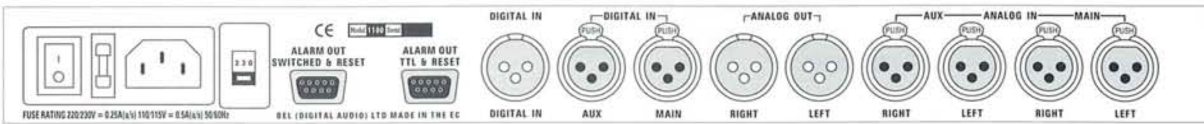
When used under serial control the Bel 6110 provides VTR emulation and supports Sony protocol. Standard stereo memory is 42 secs (1 min 24 secs mono) with 1 min 24 secs and 2 mins 48 secs options available. The Bel 6110 is a unique production tool that has become renowned for its cost saving efficiency and ease of operation.

Remote Control

SPECIFICATIONS - MODEL 6110	
Analog	
Frequency response	20Hz - 20kHz ±1dB
Dynamic range	96dB
Distortion	Less than 0.015% at +8dBm/1kHz
Audio inputs	Electronically balanced
Audio input level	+8dBm nominal
Audio input impedance	30kΩ
Conversion	16 bit Sigma Delta
Audio outputs	Electronically balanced
Audio output level	+8dBm nominal, +18dBm max
Audio output impedance	100Ω
Digital	
Digital input/output	Standard AES/EBU
Sampling frequency	48kHz
Reference	1. Internal crystal 2. Audio In 3. External word clock (48kHz) 4. External AES/EBU (48kHz) 5. External video (1V pk-pk)
General	
Timecode input	SMPTE, electronically balanced
Remote serial interface	RS422
Power requirements	110/220VAC 50/60Hz
Power consumption	35W
Dimensions	482w x 44h x 340d
Net weight	4.5kg

Technical specifications subject to change without notice.

Bel 1100 Broadcast Audio Silence Monitor



The Bel 1100 is designed to indicate the absence of audio during transmission or reception. It will accept either analog or AES/EBU digital feeds and can automatically change to an alternative input source if required. The unit will detect loss of audio on either or both channels of a stereo signal and, if the situation persists for a set interval, generate an alarm condition. An alarm sounder is provided together with changeover relay contacts and TTL outputs for remote indications. In the event of a power failure the primary inputs are connected to the outputs by electromechanical means to provide fail safe. The Bel 1100 may also be used to monitor timecode with the 0-8 sec reaction time option (please specify when ordering).

SPECIFICATIONS - MODEL 1100	
Analog audio inputs/outputs	Balanced or unbalanced
Digital audio inputs/outputs	AES/EBU
(input signals are unbuffered and pass directly through the unit)	
Time set range	4 to 60 secs in 5 sec increments or 0 - 8 sec (optional extra)
Threshold set	-60dB to -10dB in 5dB increments
Alarm outputs	2 x relay contact closures, max 24V @ 500mA each 2 x TTL +ve
Remote reset	External momentary contact closure
Power requirements	110/220VAC 50/60Hz
Power consumption	35W
Dimensions	482w x 44h x 240d
Net weight	3.5kg

Technical specifications subject to change without notice.

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PROGRAMME LOUDNESS METERING

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The loudness of television programmes is a subject which statistically has maintained its position in the viewers' top ten concerns for nearly forty years. This paper combines a study of these statistics with an objective study of the whole subject of loudness. As a result, a tentative specification for a loudness meter is put forward.

INTRODUCTION

From the time of the introduction of a second television channel in the UK, funded by the revenue from advertising, there have been complaints about the loudness of commercials. A viewing public, familiar with a single channel without commercial breaks, was split in its reaction to the new service. While a proportion of these viewers thought the new service, with its catchy jingles and potted cameos, to be fun, another quite significant proportion disliked the interruptions to continuous programming and considered the commercial breaks to be a rude intrusion of their viewing and listening.

To a certain extent these attitudes are still present but criticism has been extended to cover promotional material and certain types of light entertainment programming.

RELEVANT FACTORS

It is difficult to list definitive reasons why this televised material appears to be louder than the programming either side of it because of the subjective nature of the problem. However, a number of factors may be relevant:

1. A commercial with a punchy musical background and forceful speech will appear subjectively louder when it follows a predominantly speech based programme like the news or a drama production.

2. A particular commercial may not be liked by some viewers and the sound levels may then appear to be objectionably high.

3. Orchestral music explores the complete range of sound levels from very quiet passages to very loud ones and often there are pauses between notes or phrases. The human ear determines an average sound level for this music. Music accompanying commercials is often compressed so that the range of sound levels is limited to, usually, near the top end of the loudness scale. This results in a higher average sound level being detected by the ear.

4. Processing techniques which modify the attack and decay times may also be used.

5. It is the advertiser's intention that a commercial should be noticed (and made to stand out from the material either side of it) so that the product or service is sold, thereby making the investment in the commercial worthwhile.

6. The viewer's reaction to what he or she hears may be determined by mood. Also, the television sound may be pushed into the background because of the need for conversation or reading, etc. Any increase in loudness then becomes obtrusive.

RECOMMENDED PEAK LEVELS

The actual sound level heard by the viewer is ultimately determined by the setting of the receiver's volume control. The ratio of levels as measured on a ppm is maintained but the scale is magnified by a factor determined by the volume control setting. It is important that a fairly constant average sound level is maintained so that the viewer does not have to keep adjusting the volume control. This should also be the requirement when changing between channels.

In an effort to maintain this fairly constant average sound level the Independent Broadcasting Authority (IBA) and Independent Television (ITV) companies drew up a table of recommended peak levels for different types of programme material. This was published in the Technical Regulations, "to ensure that the subjective volume is consistent with the transmitted material whilst at the same time preventing excessive loudness changes", and is reproduced in Table 1.

Material	Normal Peaks	Full Range
Speech		
Talks, News, Drama, Documentaries, Panel Games, Quiz Shows, Announcements	5	1 - 6
Music		
Variety, Dance Music.	4.5	2 - 6
Brass Bands, Military Bands.	4	2 - 5
Orchestral Concerts.	-	1 - 6
Light Music.	5.5	1 - 6
Pop Music.	5	2 - 5
Programmes containing a high degree of compression	4	2 - 4
Commercials		
Commercials containing a high degree of compression	4	2 - 4

Table 1. Recommended Levels for Programme Material

A steady, though small, number of complaints to the IBA from viewers showed that, although every effort was made to control levels within these ranges, there was still a problem. It was obvious that the ppm was not a suitable instrument for measuring subjective loudness, so tests were set up to find an alternative solution.

THE IBA TESTS

A domestic television receiver was set up in an acoustically treated quality monitoring room and its audio output was fed to good quality loudspeakers. The volume control was adjusted for a comfortable listening level of programme sound. A high quality Bruel and Kjaer microphone was placed in the listening position and its output was taken via a measuring amplifier with 'A ' weighting to a pen recorder.

A test transmitter set to the same channel frequency was fed to the receiver and the output of an audio oscillator was adjusted to give 50kHz deviation for a peak signal level of +8dB with pre-emphasis in. Measuring amplifier gain and pen recorder sensitivity were adjusted so the full dynamic range could be accommodated on the paper trace and calibration marks were made at 4dB intervals corresponding to ppm markings. Paper speed was adjusted so that a minute of programme occupied 2 cm of trace, which ensured that all transients could be recorded.

The off-air signal was restored to the receiver and pen recordings were made of typical periods of broadcasting. VHS video tapes were also recorded so that interesting parts of trace could later be related to actual programme material.

The results showed that sound levels, which were judged as being too loud or louder than other programme materials, were indicated as being so on the recorded paper trace. This did not necessarily correspond to any peak indication but more to the density of ink on the paper, suggesting that the problem is associated with average sound level or sound density with time. Promotions, company idents and signature tunes during titles and credits were similar to the commercials in causing loudness problems. The problems were not restricted to the commercial channels but occurred on all four UK terrestrial channels.

These finding agreed very well with those of Thames Television, who had taken the investigation a stage further and were considering the design of a loudness meter based upon the operation of the human ear.

AUDIENCE RESEARCH

A recent (end of 1993) survey carried out by the British Audience Research Board (BARB) considered sixteen attitude statements on the subject of television advertising, using a panel size of around 3,000 for the week in question. Four items relevant to sound levels asked respondents to agree or disagree that television advertisements:

1. are louder than programmes
   -more than one in two agreed
   -less than one in five disagreed
2. are broadcast with excessive sound levels
   -more than four in ten agreed
   -more than one in five disagreed
3. make me turn the sound down
   - nearly four in ten agreed
   - a similar proportion disagreed
4. are too noisy
   - around one in three agreed
   - one in four disagreed

The remainder of the questions dealt with other aspects of television advertising but there was clear evidence that substantial proportions of viewers were aware of sound levels in advertisements which are seen by many to be excessive.

On the basis of this survey and the previous work carried out by the IBA, the new Independent Television Commission (ITC) decided to place a contract with Thames Engineering to finalise the design of a loudness meter and produce a number of prototypes. The prototypes are to be used for an experimental period by ITV, Channel 4 and the ITC and it is hoped standard production units will be available soon afterwards.

DEFINING THE PROBLEM

Having established that there is a problem, which viewers have described as excessive loudness, we need to be sure that this loudness is the same as that which we understand by the term in the scientific sense. If we provisionally define the problem as one of differential sensory loudness between short programme segments, the assumption can be tested subjectively. As it turned out, this subjective test stage did much more than just confirm this assumption and, to see how this came about, it is worth analysing some of the opinions, which were held over the many years that the loudness problem had existed. Foremost among the opinions was that production houses used vast amounts of dynamic range compression in order to produce high levels of effective loudness on commercials, whilst maintaining the recommended peak modulation levels.

Two factors soon killed this theory. Firstly, independent sound operators could often confirm that no compression had been used on problem items, and secondly, excessive loudness of promotion materials had been found on excerpts, which had been produced by in-house teams. Clearly, dynamic range, like peak modulation levels, contributed to the apparent loudness but was by no means the full story. A second opinion, which was widely held, was that the loudness factor was purely 'perceptual' that is, in the mind of the individual. This idea should have been discounted by the complaints statistics, but it was nevertheless a vital item for the subjective test, because only sensory loudness can be metered. Any significant perceptual element will therefore reduce the effective value of loudness metering.

THE SUBJECTIVE TEST SERIES

As far as the 40 families who took part in these tests knew, the video tape consisted of five bands of five 30 second promotions and commercials. Each group of excerpts was interspersed with programme material representing a wide range of interests. Each family member, of which there were just over 120 in total, filled in a form using a CCIR 5 point scale of loudness for each of the 25 numbered sections.

In reality there were two tapes with different assembly orders, thus producing two different perceptual conditions, and there were also two items numbered 23, in order to test attentiveness.

The results, summarised in Figure 1, were quite encouraging. The results for both tapes identifeid the same three statisitcally 'quiet' items and the same four statistically 'loud' items, albeit one marginally so. Sensory loudness had won by a large statistical margin over perceptual, so, in theory, it should be possible to build a meter to measure it. The next question was how?

DESIGNING A LOUDNESS METER

Defining the problem in scientific terms as differential sensory loudness variation between short excerpts of different audio content, seems to have been close enough to the truth for the statistical tests to agree. The next stage therefore is to list the quantities which contribute to sensory loudness, and design a meter which incorporates as many of these parameters as possible.

One obvious unknown, at the start of the project, was the preferred home listening level.

The Thames Television Playout Centre of the late 1980's used Digital VTR sound on the MII format, combined with NICAM distribution and transmission. This enabled us to be sure of the levels and frequency response throughout the system all the way to the home. The actual mean sound level in the home, however, is always at the control of the viewer.

A series of tests was therefore done in a simulated home environment, and a consistent level of 68 phon (±3phon) was established for non-experts. This value reconfirmed the 70 phon level found by CBS in the early 1970's, but an interesting side issue was that the expert sound operator's preferences varied much more widely, covering the range from 63 and 82 phons.

68 phon is the loudness of a 1kHz tone at 68db SPL. The loudness of other frequencies at 68dB SPL varies with the frequency, and this is the first important characteristic to be built in to a loudness meter. This characteristic alone produces a sound level meter type of response, ideal for wideband noise, but poor for music and speech the two characteristics which distinguish music from speech and noise are spectral content and dynamics, and so these two extra factors must be built in to a loudness meter.

A final requirement is that the display must be acceptable to a sound operator. That is a complex subject, not the least because of the wide variation in programme level meters at present used throughout the world. Since loudness is referred to that of a 1kHz tone, the solution chosen here was to produce a quasi - 1kHZ tone output from the electronics. This is then equivalent to the sensory loudness value, and it can drive any conventional programme meter to give a loudness response. Figure 2 shows the block diagram of this loudness drive circuit.

STANDARDISATION OF A LOUDNESS METER

The four technical factors which need to be standardised are:-

1) The frequency response to a single frequency.

2) The summation factor for wide band signals.

3) The dynamic response to which:-

4) Preferred display characteristics could be added.

Looking at existing standards as sources of each of these four factors, ISO and IEC publications cover items (1) and (4) more than adequately. Number (2) is covered in part by ISO 532, which describes a non-real time graphical method of calculating the loudness of complex sounds. Taking a pragmatic view of this requirement, it may be adequate to simply give an overall response to a quazi-random 'pink' noise source. The only trouble with this approach being the need to specify the noise source! Factor (3) is left, therefore, as the only unknown. Here again there are precedents in the psycoacoustic models used for bit rate reduced audio coding but, for completeness, a simple subjective test can be performed by comparing artificial or real pulsed audio sources with a 1kHz reference tone (Fig.3). The well known tracks from the EBU SQAM disc, "claves" and "castanets", have proved ideal in this context and Figure 3 shows a typical subjective test set up.

CONCLUSION

By a process of analysis and subjective testing at each stage along the way, we hope we have at last come to some scientific explanation of the loudness variations experienced by television listeners. The important stage of subjective testing the meter specifications has yet to be done but, in contrast to the situation only a few years ago, at least we can be certain that an economic electronic solution can be found to this specification. Therefore, such meters could be rapidly made available, not only within television production areas, but for evaluation within the wider audio field as well.

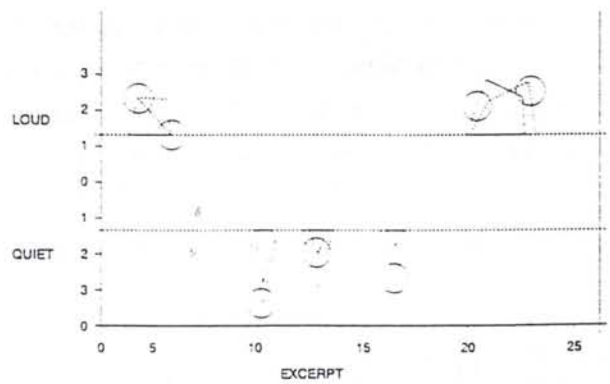

FIGURE 1
Subjective loudness grades for 25 Excerpts on two tapes

Grades - 1 = Perceptable but not annoying
2 = Annoying
3 = Very annoying

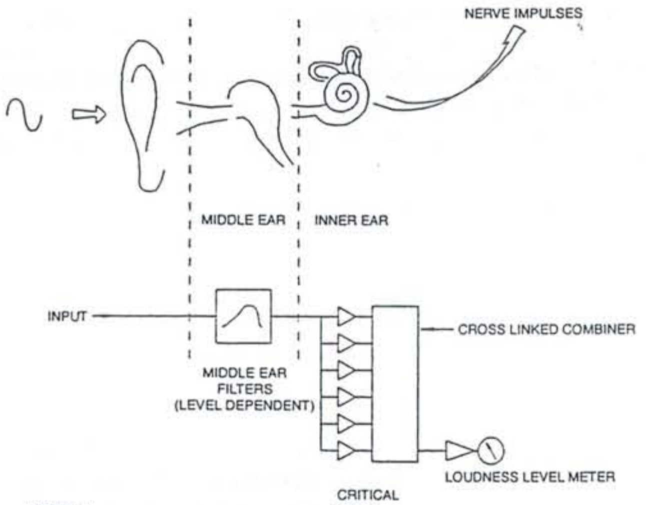

FIGURE 2
An electrical Analogue of the Ear

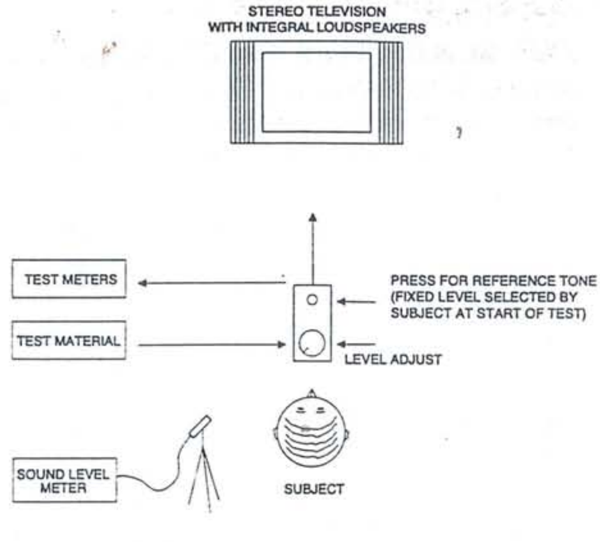

FIGURE 3
Quantative Tests Arrangement

UPDATE TO AES PAPER, "PROGRAMME LOUDNESS METERING", OCTOBER 1996

Since the paper, *Programme Loudness Metering* was presented by Emmett and Girdwood at the 1994 AES UK Conference, "Managing the bit budget", three prototype loudness meters have been built and tested extensively by ITV and Channel 4 television companies in the UK. This update is intended to carry on from the text in the paper. say what has happened in the intervening time and what the situation is in October 1996.

The prototype meters take left and right audio inputs and derive the mono signal and the loudness drive signal. One meter displayed the loudness signal levels on an LED readout and two meters were installed in modified in-vision displays showing the ppm levels at the left of the picture and the mono and loudness levels on their right. The units are intended as monitoring meters only and do not control audio levels but the output is available to drive a VCA for automatic control if this is required.

They were installed for varying lengths of time in a number of television station locations where there was a recognised need for loudness monitoring and control. These included production galleries and post production suites. bar coding stations in VT areas where programmes and commercial breaks are put together for automatic playout. commercial dubbing areas and transmission suites. In all locations. the meters were considered to be a useful addition to the monitoring facilities and of more use than traditional meters in that the indicators did confirm the subjective impression of loudness.

Another use was found in the areas of film transfer and playout. Sound tracks on feature films are made for theatrical presentation and can have a very wide dynamic range. The meters were used to monitor loudness during range compression to give a sound track more suitable for television in the home. They were also satisfactorily used as a calibration tool while setting up levels on audio processors during experiments to control transmission outputs.

These trials have shown that the loudness meter does achieve what was intended as it is now being produced by Chromatec Video Products Ltd and distributed by Michael Stevens and Partners. Invicta Works. Elliott Road. Bromley. Kent. BR2 9NT.

Interest in the meters was shown at IBC 1994 in Amsterdam and after a demonstration to members of the EBU group concerned with studio origination equipment. Unfortunately, in re-organsation at the EBU this group has been disbanded. but now that the meter is in production the matter of recommendation by the EBU and subsequent standardisation should be pursued.

The ITC Technical Performance Working Party has listed possible improvements to the original design that might be considered in the future. They are:

1. A new scale not numbered like a ppm. This may have alphabetic characters or musical loudness terms.
2. A line-up level that corresponds to PPM 5. not PPM 4.
3. A non-linear or "magnified" scale that gives a better indication of obvious loudness or what is / is not acceptable.
4. A peak hold facility to indicate the previous maximum level (present on in-vision meter but not on LED scale).

As a nice ending to this update and a confirmation of the effectiveness of the loudness meter it is now possible to say that one of the companies involved in the trials has placed an order for about 20 of the first production units.

Chas Girdwood. Project Manager (Studio Development). ITC. October 1996

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