session 2007

durée 3 heures
coefficient : 2

BREVET DE TECHNICIEN SUPERIEUR DES METIERS DE L'AUDIOVISUEL

OPTION METIERS DU SON

EPREUVE : TECHNOLOGIE DES EQUIPEMENTS ET SUPPORTS

Ce sujet comporte **17** pages.

LISTE DES DOCUMENTS :

Annexe 1 : Midas heritage 3000 input module

Annexe 2a, 2b, 2c : Studer D19m stagebox & D19m digital system components

Annexe 3 : AKG WS4000 frequency map

Annexe 4 : AKG C5900

Annexe 5 : D-LINK DES-1228 24 Port switch

Annexe 6 : RTS ADAM CSedit user manual

Annexe 7 : Studer D19 master sync generator

Annexe 8 : SRW-5000 Specifications

Annexe 9 Dolby E

Avertissement : l'emploi de tout document est interdit, mais celui d'une calculatrice conforme à la réglementation en vigueur est autorisé.

PRESENTATION DU THEME D'ETUDE :

La société de prestation pour laquelle vous travaillez est consultée pour effectuer la captation multi caméra d'un spectacle en vue de la réalisation d'un DVD. Cette captation s'effectuera au cours de 2 représentations du spectacle à l'aide d'un car régie et certains choix d'équipements ou de méthodes restent à faire.
Le spectacle est sonorisé à l'aide d'une console Midas héritage 3000.
Des micros HF system WS4000 AKG sont mis en oeuvre dans le spectacle.
La captation se fera en vidéo numérique 4 :2 :2 SD mais la production souhaite archiver une version HD pour une éventuelle commercialisation future en HD-DVD™ ou blu-ray Disc™.

De ce fait le mélangeur du car régie (XTEN DD HD) a été choisi pour sa capacité à gérer simultanément un flux vidéo HD et SD (simulcast).

L'enregistrement vidéo est confié à des magnétoscopes au format DIGITAL BETACAM pour la SD et on envisage d'enregistrer la HD avec des magnétoscopes au format HDCAM-SR.
Le son de la captation est géré par une console Studer D950 avec en premix une console YAMAHA DM1000.
Les différentes sources sonores sont « splittés » (vers la sonorisation façade, et le car régie) et récupérées depuis la scène par une « stagebox » D19m Studer.
Les ordres entre les différents techniciens sont gérés par une station RTS ADAM matricielle et comportent une partie sans fil.

1. **Alimentation électrique du car régie :**

   Le car est alimenté en triphasé 380 volts avec une consommation de 65 kVA.

   1.1. En cas de premier défaut électrique sur l'installation (courant de fuite par exemple) l'alimentation du circuit concerné ne disjoncte pas afin d'assurer la continuité du direct. Le défaut est par contre signalé par un Contrôleur Permanent d'Isolement. Quel est le régime de neutre utilisé pour cette installation ?

   1.2. Quel autre régime de neutre est plus couramment utilisé sur des installations domestique ?

   1.3. Dans ce cas quels dispositifs permettent la coupure du courant dès le premier défaut détecté ?

2. **Console de sonorisation façade MIDAS heritage 3000**

   L'annexe 1 montre le synoptique d'un module d'entrée de cette console

   2.1. Expliquer le terme « four band fully parametric equaliser »

   2.2. Quel est le rôle de l'option en entrée « transformer balancing » ?

   2.3. Donnez la signification du terme VCA.

   2.4. Quel est le rôle du VCA dans ce module ?

3. **Stagebox D19m Studer (annexe 2) :**

   Cette stagebox est modulable en fonction des besoins. Une présentation générale est donnée à l'annexe 2a. Les annexes 2b et 2c décrivent les différents modules utilisables.
   La stagebox est reliée à la console D950 par une interface MADI

3.1. Donnez les principales caractéristiques de l'interface MADI (type de liaison, nombre de canaux, fréquence d'échantillonnage, résolution, support physique).

3.2. A partir des annexes 2b et 2c décrire les fonctionnalités des modules « D19m MP4RC », « D19m C4AD/24 » et « D19m MADO ».

3.3. En déduire le nombre de modules nécessaires à l'acquisition, la conversion et le transport des 36 canaux récupérés depuis la scène.
Comment sont distribuées les sources vers le car régie et vers la console de sonorisation MIDAS H 3000 ?

3.4. La description du module D19m MADO fait apparaître que ce dernier peut recevoir une synchronisation externe. De quel type de synchronisation s'agit-il et quel est son rôle ?

4. **<u>Choix des microphones HF : (annexe 3 et 4)</u>**

Le spectacle utilise 8 microphones de type HF.

4.1. Rappelez le type et les paramètres de la modulation utilisée par les micros HF.

4.2. Le système HF utilisé est le WMS4000 de AKG qui permet un large choix de canaux de fréquence des émetteurs/ récepteurs. Les fréquences possibles sont données à l'annexe 3. Ces fréquences peuvent être perturbées par les émissions de télévision de l'émetteur local résumées dans le tableau ci-dessous.
Quelles sont les fréquences du système WMS 4000 à ne pas utiliser.

CANAL	Fq Mhz	Type d'émetteur
21	470-478	*analogique*
24	494-502	*analogique*
27	518-526	*analogique*
29	534-542	*TNT*
30	542-550	*TNT*
32	558-566	*TNT*
35	582-590	*TNT*
36	590-598	*TNT*
51	710-718	*analogique*
53	726-734	*analogique*
54	734-742	*analogique*
65	822-830	*analogique*

Un des micros utilisé est un micro main avec émetteur HT4000 et capsule C5900WL dont les caractéristiques (identiques au C5900 de la même marque) sont détaillées à l'annexe 4)

4.3. Expliquez les 4 caractéristiques suivantes :
"Sensitivity", "sound pressure level for 1% (3%) THD", "transducer type", "polar pattern"

4.4. Le micro capte une source acoustique d'un niveau de 114 dBspl. Quelle est la tension à vide fournie par ce micro ?

4.5. La liste des caractéristiques annonce une impédance de 200 ohm. De quelle impédance s'agit-il ?

5. Informatique : (annexe 5)

La partie son du car régie est dotée d'un ordinateur de type PC avec différents logiciels permettant d'éditer des sons ou de les utiliser lors d'un direct (Sound Forge, Live). Ce PC comporte une carte son DIGIGRAM PCX1221HR dont quelques caractéristiques sont énumérées ici :

- sound card with 1 AES/EBU input and 6 AES/EBU outputs.
- Short-length PCI format (only 175 mm/6.875 inches long)
- 24-bit/192 kHz converters
- 66 MHz/64-bit PCI interface
- extremely powerful DSP

5.1. Expliquer "66 Mhz/64 bits PCI interface" : calculez le débit de cette interface, sachant qu'un échange a lieu tous les 2 cycles d'horloge.

5.2. Quel est l'intérêt d'un DSP ?

5.3. Le PC est équipé de mémoires DRAM DDR PC3200 400 MHz pour une capacité de 1 Go. Expliquez les termes DRAM, DDR et 400 MHz

Ce PC est relié à tous les autres PC du car régie par l'intermédiaire d'un commutateur (switch) D-LINK DES-1228 (ANNEXE 5).

5.4. Quel est l'intérêt d'utiliser un commutateur plutôt qu'un concentrateur (HUB) ?

5.5. Que signifie le terme « 10/100/1000T gigabits Port » ?

5.6. L'administrateur réseau a attribué une adresse IP au PC de la régie son. Qu'est-ce qu'une adresse IP ?

5.7. Quelle est la différence avec l'adresse MAC ?

5.8. Quel est le type de réseau constitué dans le car régie ? Quelle est sa topologie ?

6. mise en place des réseaux d'ordre (annexe 6)

*Pour gérer la communication entre tous les intervenants du spectacle et ceux de la captation (techniciens, équipe de production...) vous êtes chargé de configurer un système RTS ADAM 80*80 (ADAM = Advanced Digital Audio Matrix).*

6.1. Tous les cadreurs ainsi que le réalisateurs doivent pouvoir communiquer ensemble sur la même ligne. Chaque opérateur est entendu par l'ensemble des autres personnes connectées. Quel terme désigne ce mode de communication ?

6.2. Certaines communications doivent pouvoir se faire entre seulement 2 intervenants (par exemple directeur de production et réalisateur, chef d'équipement et ingénieur de la vision). Ce mode s'appelle ISO Channel.
Dire et justifier si l'on peut obtenir ce fonctionnement avec un système d'intercom à 2 fils ?

6.3. L'annexe 6 propose un extrait du mode d'emploi du logiciel de programmation du système ADAM.
A l'aide de ce document expliquer le mode de fonctionnement IFB.

6.4. Dans le cas de cette captation quelle signal audio devez vous considérer comme étant« program » ?

7. **Enregistrement de la vidéo :**

Les magnétoscopes initiaux embarqués dans le car régie sont des Sony DVW-500P au format DIGITAL BETACAM

7.1. Dans ce format quelles sont les caractéristiques du signal vidéo enregistré (structure du signal vidéo, résolution, taux de compression)?
7.2. De même quelles sont les caractéristiques du signal audio ?
7.3. Sur Le panneau arrière on trouve une entrée « ref video input ». Dessinez sur votre copie le chronogramme de ce signal en indiquant en abscisse les durées caractéristiques et en ordonnée les tensions typiques.
7.4. Lors de l'enregistrement d'un signal audio issu des entrées analogiques du magnétoscope on peut appliquer un réglage dénommé « pre-emphasis ». Quel est le rôle de ce réglage ?

8. **Synchronisation audio numérique :(annexe 7)**

Les signaux mixés par la table de mixage Studer D950 sont enregistrés sur trois magnétoscopes DVW-500P, mais ceux-ci ne disposent pas d'entrée wordclock.

8.1. Quel est le rôle d'un signal « wordclock » ?
8.2. Donnez ses principales caractéristiques.
8.3. A l'aide de l'annexe 7 décrivant le générateur de synchronisation de la table de mixage Studer D950 expliquer les liaisons à mettre en oeuvre pour asservir cette dernière aux équipements vidéo du car régie.

9. **Enregistrement HD : (annexe 8)**

On étudie la possibilité d'enregistrer la captation en HD à l'aide de magnétoscope SONY SRW-5000 au format HDCAM-SR dont les caractéristiques sont résumées à l'annexe 8

9.1. D'après ce document décrire le signal vidéo enregistré par ce magnétoscope et le comparer à un signal vidéo numérique SD.
9.2. Le HDCAM-SR supporte aussi bien le 720P que le 1080i. Quelle différence faites vous entre ces deux résolutions ?
9.3. Donnez les caractéristiques de l'audio enregistré par ce magnétoscope.
9.4. En quoi le choix du SRW-5000 serait-il intéressant pour cette captation, d'un point de vue audio ?
9.5. Donnez le nom et le débit de l'interface capable de transporter l'ensemble des signaux enregistrés ou lus par le magnétoscope.

10. **Post production : (annexe 9)**

10.1. Les opérations de mixage ont été réalisées en conservant une quantification de 24 bits. En vue de la réalisation du DVD, on vous demande de réduire cette quantification à 16 bits. Pour ce faire vous appliquez une fonction « dither ».
Expliquez en quoi cela consiste et justifiez son utilisation.

Un mixage multicanal est également à fournir. Il a été décidé que ce mixage serait encodé en Dolby AC-3 (débit de 384 kbits/s à 48 kHz, 16 bits).

10.2. A l'aide d'un schéma montrez la répartition spatiale des canaux du Dolby AC-3.

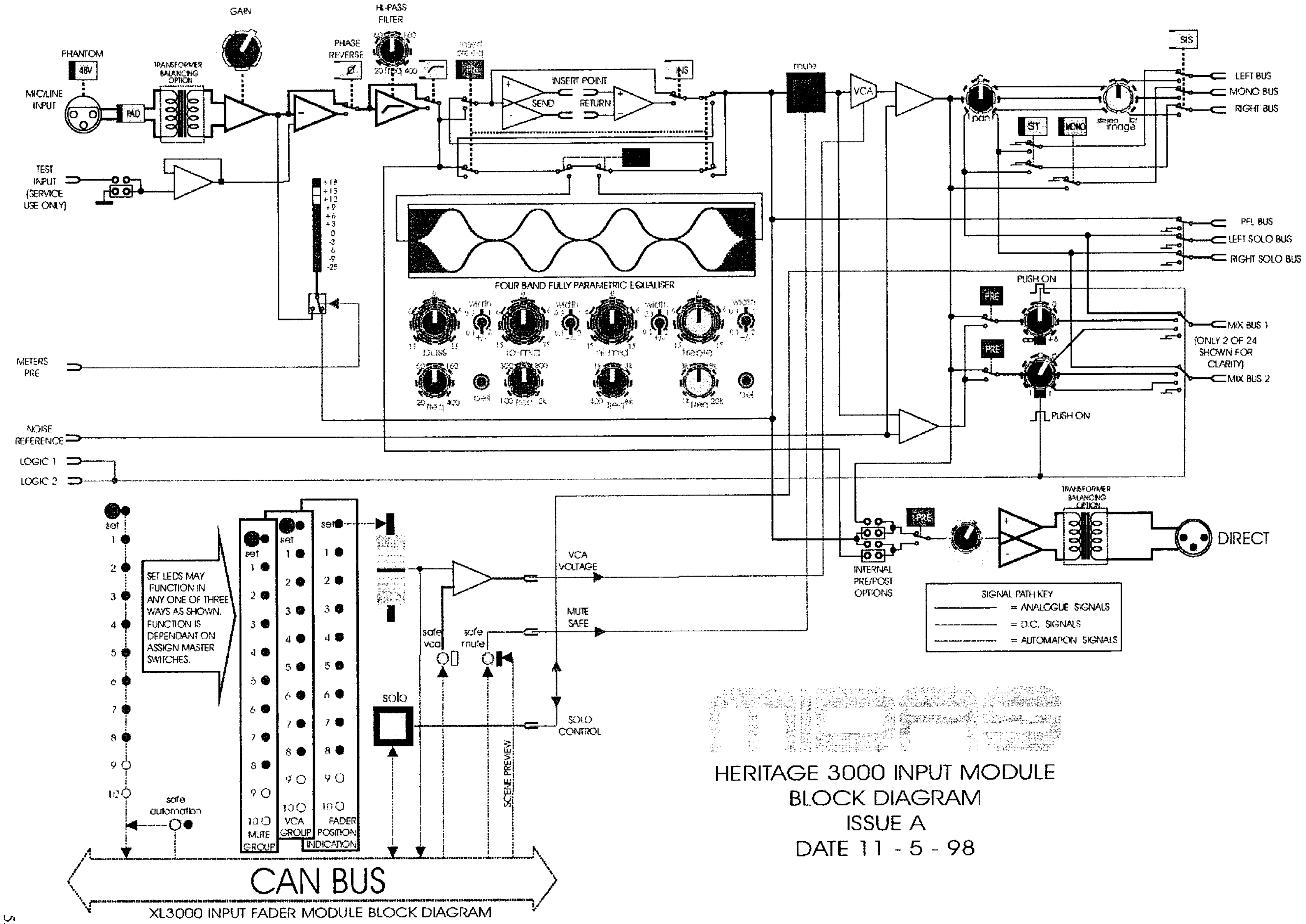
10.3. Calculez le taux de compression appliqué lors de ce codage.

Le car régie de captation était doté d'un encodeur Dolby E DP571. L'annexe 9 est une description de ce type de codage.

10.4. A l'aide de l'annexe 9, expliquez les différences fondamentales entre le Dolby AC-3 et le Dolby E. (type de codage, applications...)

10.5. Aurait il été intéressant d'utiliser ce codeur lors de la captation ?

Annexe 1

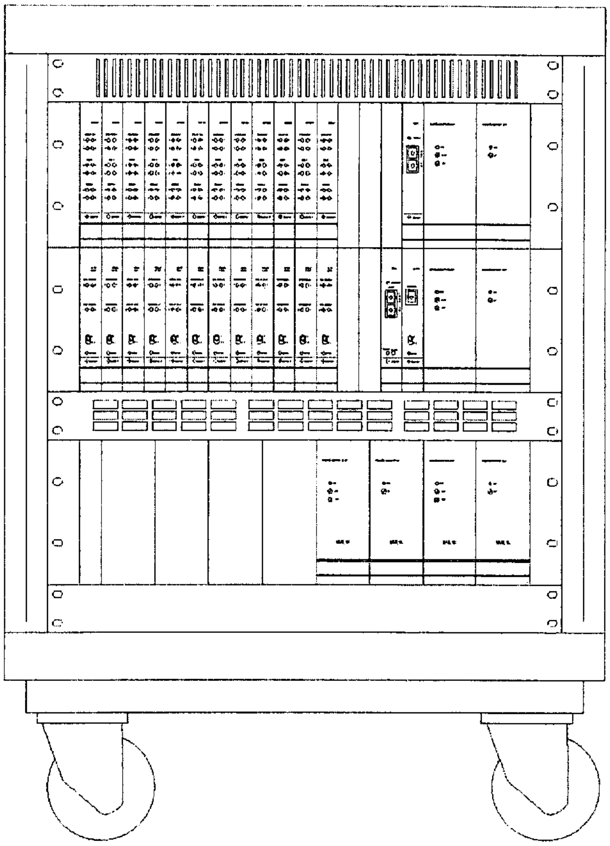

GAIN
HI-PASS FILTER
PHANTOM
48V
TRANSFORMER BALANCING OPTION
PHASE REVERSE
MIC/LINE INPUT
PAD
INSERT POINT
SEND
RETURN
INS
mute
VCA
LEFT BUS
MONO BUS
RIGHT BUS
pan
ST
MONO
TEST INPUT (SERVICE USE ONLY)
PFL BUS
LEFT SOLO BUS
RIGHT SOLO BUS
FOUR BAND FULLY PARAMETRIC EQUALISER
PUSH ON
PRE
MIX BUS 1
(ONLY 2 OF 24 SHOWN FOR CLARITY)
MIX BUS 2
METERS PRE
NOISE REFERENCE
LOGIC 1
LOGIC 2
DIRECT
INTERNAL PRE/POST OPTIONS
SIGNAL PATH KEY
= ANALOGUE SIGNALS
= D.C. SIGNALS
= AUTOMATION SIGNALS
SET LEDS MAY FUNCTION IN ANY ONE OF THREE WAYS AS SHOWN. FUNCTION IS DEPENDANT ON ASSIGN MASTER SWITCHES.
set
MUTE GROUP
VCA GROUP
FADER POSITION INDICATION
VCA VOLTAGE
MUTE SAFE
safe vca
safe mute
solo
SOLO CONTROL
SCENE PREVIEW
safe automation
CAN BUS
XL3000 INPUT FADER MODULE BLOCK DIAGRAM
HERITAGE 3000 INPUT MODULE
BLOCK DIAGRAM
ISSUE A
DATE 11 - 5 - 98

Annexe 2a

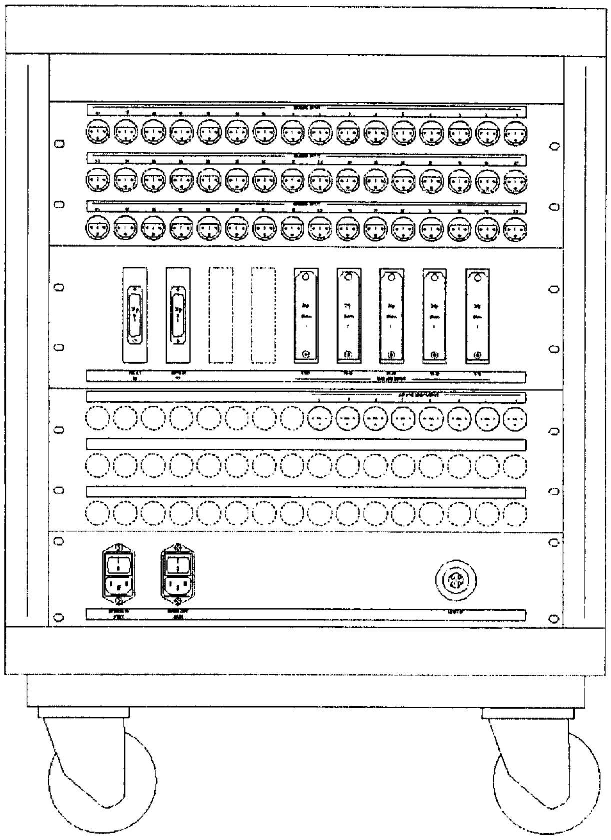
13.6 Studer D19m Stagebox

(only one of the many possible variations shown)

Front view:



Rear view:



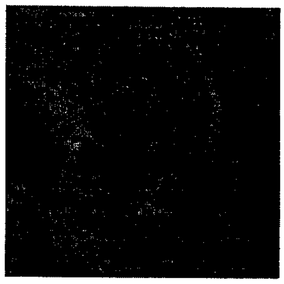
A transportable stagebox can be ordered as an option for the system. It typically consists of one or several Studer D19m frames in a flightcase. The number of frames and cards is variable and is usually done to customer specification. The following are some of the characteristics of the Studer D950 Stagebox system:

- A D19m frame can hold input and output cards (mixed arrangement).
- The connection to the D950 is typically via a single, robust, four-core optical fibre cable for MADI in/out, control receive/transmit signals, as well as the sync. This cable can cover distances of up to several hundred meters.
- The control signals are standard RS232/RS422 connections.
- A low-noise fan is required for large stageboxes.
- Optional redundant power supply is available (as shown in the above example).

Annexe 2b

Studer D19m Digital System Components

This range of digital modules can be used in stand-alone applications or for expanding the interfaces of a Studer digital product. The Studer D19m range includes two mounting frames both of which are equipped with power supplies. For further details on these racks and series of cards, please refer to the Studer D19m brochure.



D19m ADATI

Dual 8-channel ADAT Input

Optical Receiver and TDM Bus Driver, converting 2 x 8-channel ADAT* inputs into 16 time slots on the TDM Bus and eight AES/EBU stereo outputs. Optionally, two input channels can be transferred to AES/EBU stereo outputs. Sync by the frame signal of the TDM Bus; in standalone applications via an AES/EBU sync input or directly from the optical input.

D19m ADATO

Dual 8-channel ADAT Output

Optical Transmitter, converting 16 time slots on the TDM bus to two 8-channel ADAT outputs. The card can also be equipped with eight AES/EBU receivers for stand-alone applications.

D19m MADI

MADI Input for fibre optic/coaxial cable

MADI Receiver and TDM Bus driver, converting a MADI frame into 56 TDM Bus time slots . The TDM Bus is synchronized by the received sync signal. In slave mode, the board is synchronized by a back plane signal. The fiber-optic cards extract the sync signal from MADI (no additional sync input needed). External control via an RS 485 interface. Optional redundant MADI input / through with automatic switcher.

D19m MADO

MADI Output for coaxial cable

TDM Bus Receiver and MADI transmitter, converting up to 56 time slots into a MADI frame. The time slot allocation between the TDM signals and the MADI frame can be externally controlled via an RS 485 interface. This function allows the configuration of a very simple 56 x 56 MADI router. The TDM Bus is synchronized by the received sync signal. In case of a missing sync signal, the unit automatically generates a high-precision sync signal.

D19m TDIFI

Dual 8-channel TDIF Input

TDIF** Receiver and TDM Bus Driver, converting 2 x 8 channel TDIF inputs into 16 time slots on the TDM Bus and optional eight AES/EBU stereo outputs. Sync by the TDM bus; in stand-alone applications via AES/EBU sync input or directly from the TDIF input.

D19m TDIFO

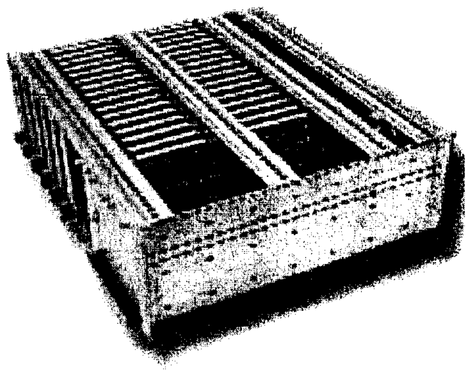
Dual 8-channel TDIF Output

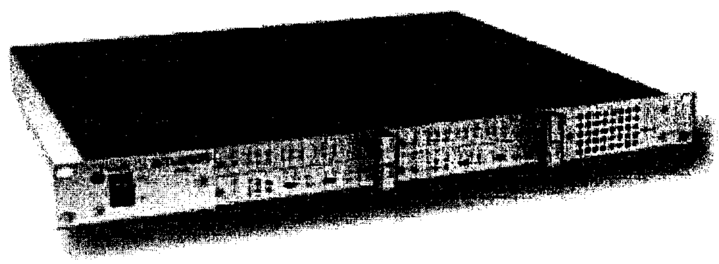
TDIF Transmitter, converting 16 time slots on the TDM bus to two 8 channel TDIF outputs. The card can also be equipped with eight AES/EBU receivers for stand-alone applications.

D19m frames

One 19" 1 U unit accommodates up to four D19m cards, with power supply

Three 19" 3 U units accommodate up to 16 D19m cards (14 I/O interface cards, up to two MADI I/O cards), with power supply

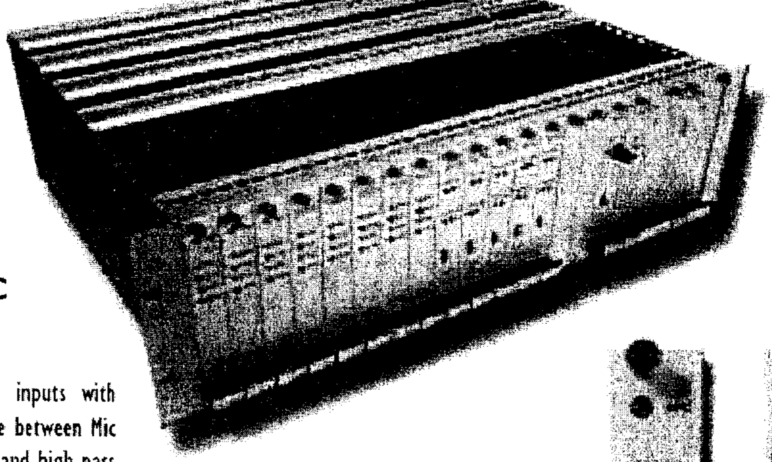




**TDIF is a trademark of TEAC

*ADAT is a trademark of Alesis

Annexe 2c



D19m MP4RC

Quad Remote Controlled Mic/Line Input

Four transformer-balanced Mic/Line inputs with gain control in 1 dB steps. Switchable between Mic and Line level. 48 V phantom power and high pass filter for Mic/Line inputs. Additional split output per Mic/Line input. One external mute (GPI) input per Mic/Line input. Electronically balanced outputs. Remote controlled from the D19m Remote Controller card. Clip Protection for all four Mic/Line inputs in common.

D19m RCC

Remote Controller Card for MP4RC Mic/Line Input

Remote controller card for the D19m MP4RC Quad Remote controlled Mic/Line input. The card is controlled via a serial interface (RS 422 with 38.4 kBd, standard 9-Pin on backplane or optical IF on front). Eight signaling IN and eight signaling OUT via GPI, remote controlled via serial IF. AES/EBU Synch distribution (four outputs) optionally mounted.

D19m C4AD/24

Quad 24-bit A/D Converter

Stand-alone module, converting four analog inputs to two AES/EBU outputs. TDM Bus Driver, converting four analog inputs to four time slots in the TDM Bus. 24-bit Delta-Sigma Conversion. Sync external or internal; automatic switch over to internal source in case of missing external sync signal.

D19m C4AD NS/24

Quad 24-bit A/D Converter with Noise Shaping

Stand-alone module, converting four analog inputs to two AES/EBU outputs. TDM Bus Driver, converting four analog inputs to four time slots in the TDM Bus. 24-bit Delta-Sigma Conversion. Sync external or internal; automatic switch over to internal source in case of missing external sync signal.

D19m C4DA/24

Quad 24-bit D/A Converter

Stand-alone module, converting two AES/EBU inputs to four analog outputs. TDM Bus Receiver, converting four time slots in the TDM Bus to four analog outputs. 24-bit Delta-Sigma Conversion. External sync input; automatic switch over to one of the AES/EBU inputs in case of missing external sync signal. Special electronically balanced output circuit, providing functionality similar to a balanced floating output.

D19m AESI

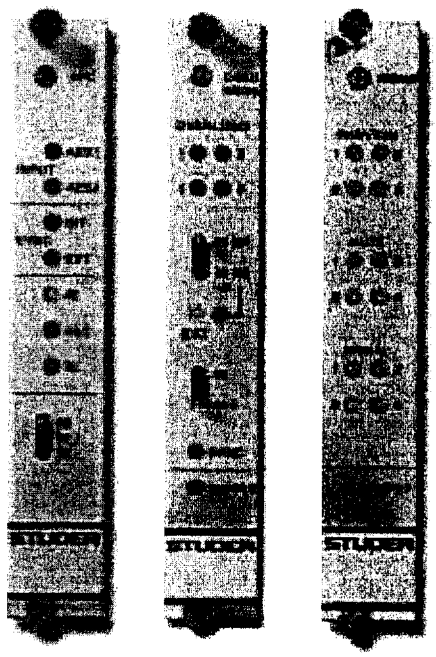
Dual AES/EBU Input

TDM Bus Driver, converting two AES/EBU inputs to four time slots in the TDM Bus. Channel status information from the inputs transferred on the TDM Bus (transparent interface).

D19m AESI SFC

Dual AES/EBU Input with asynchronous SFC

Stand-alone module, converting two AES/EBU inputs to two AES/EBU outputs using asynchronous Sampling Frequency Converters. TDM Bus Driver, converting two AES/EBU inputs to four time slots in the TDM Bus using asynchronous Sampling Frequency Converters. Internal Sync (32, 44.1 or 48 kHz) or External Sync input; automatic switch over internal reference in case of missing external sync signal.



D19m AESO

Dual AES/EBU Output

TDM Bus Receiver, converting four time slots in the TDM Bus to two AES/EBU outputs or to four AES/EBU mono outputs depending on Channel mode setting:

MONO: Each TDM time slot is fed separately to one AES/EBU output.
STEREO: Two TDM time slots are fed as a stereo pair to two parallel AES/EBU outputs.
CHANNEL: Two TDM time slots are fed as a 2-channel pair to two parallel AES/EBU outputs.

Annexe 3

EU TV	TV CH		WMS400/4000		19 freq
Channel	low	High	Hard ware version	Frequency range	combs
43	646.000	654.000	band 1	650-680	650.000
					650.300
					650.750
					651.350
					651.725
					652.850
					653.750
44	654.000	662.000			656.975
					658.250
					661.625
45	662.000	670.000			664.400
					666.050
					667.925
					668.750
46	670.000	678.000			671.000
					673.625
					675.575
					676.100
					676.775
47	678.000	680.000	band 2	680-710	680.000
47	680-686				680.300
					680.750
					681.350
					681.725
					682.850
					683.750
					686.975
	686.000	694.000			688.250
					691.625
49	694.000	702.000			694.400
					696.050
					697.925
					698.750
					701.000
50	702.000	710.000			703.625
					705.575
					706.100
					706.775
52	718.000	726.000	band 3	720-750	720.000
					720.300
					720.750
					721.350
					721.725
					722.850
					723.750
					726.975
53	726.000	734.000			728.250
					731.625
					734.400
54	734.000	742.000			736.050
					737.925
					738.750
					741.000
					743.625
55	742.000				745.575
					746.100
					746.775

EU TV	TV CH		WMS400/4000		19 freq
Channel	low	High	Hard ware version	Frequency range	combs
57	758.000	766.000	band 4	760 - 790	760.000
					760.300
					760.750
					761.350
					761.725
					762.850
					763.750
58	766.000	774.000			766.975
					768.250
					771.625
59	774.000	782.000			774.400
					776.050
					777.925
					778.750
					781.000
60	782.000	790.000			783.625
					785.575
					786.100
					786.775
61	790.000	798.000	band 5	790-820	790.000
					790.300
					790.750
					791.350
					791.725
					792.850
					793.750
					796.975
62	798.000	806.000			798.250
					801.625
					804.400
63	806.000	814.000			806.050
					807.925
					808.750
					811.000
					813.625
64	814.000	820.000			815.575
					816.100
					816.775
66	835.000	838.000	band 6	835-862	835.000
					835.300
					835.750
					836.350
					836.725
					837.850
67	838.000	846.000			838.750
					841.975
					843.250
68	846.000	854.000			846.625
					849.400
					851.050
					852.925
					853.750
69	854.000	862.000			856.000
					858.625
					860.575
					861.100
					861.775

)*** **Example:**

If TV channel 44 is occupied by a local TV station, the frequencies
656.975 658.250 661.625 have to be excluded.
Combining band1 with band 3 yields the new frequency set: From 650 to 706.775 Mhz plus the set from 720 to 746.775 MHZ

Annexe 4

AKG
H A Harman International Company

C 5900M

EMOTION I TriPOWER Condenser Microphone

Description

The AKG Model C 5900M is a rugged, premium-quality supercardioid condenser microphone intended for on-stage vocal and instrument pickup. Its supercardioid pickup pattern is uniform with frequency and its response has been specially-tailored at mid and high frequencies to deliver excellent on-stage vocal applications. A switchable 6 dB sensitivity boost and 12 dB-per-octave bass cut below 100 Hz. provides flexibility in dealing with various input sensitivities and proximity effect. The C 5900M's condenser capsule is shock mounted with a unique rubber spider suspension and is protected by a dent-resistant, easily-removable wire mesh grille. The grille also has an internal pop filter to minimize plosives and wind noise. The smooth triangular shape of the microphone handle and its soft-touch finish makes it easy to hold. The lightweight body is die-cast zinc alloy and has a dark matte black finish for minimum reflection and glare. The C 5900M can be powered by phantom power from 9 - 52 volts or with the optional AKG B 15 battery adaptor.

Specifications

Transducer type:	Prepolarized backplate condenser
Frequency response:	20 - 20 kHz
Polar pattern:	Supercardioid
Power requirements:	9 - 52 Vdc
Impedance:	200 ohms
Recommended load impedance:	>2000 ohms
Output connector:	XLR-M
Sensitivity:	6 mV/Pa; -44 dB (re 1 V/Pa)
Switchable bass roll-off:	6 dB/octave
Switchable bass cut:	12 dB/octave
Sound pressure level for 1% (3%) THD:	139 dB SPL (142 dB SPL)
Environmental operation:	
Temperature:	-10°C to +65°C
Maximum relative humidity at +20°C:	90%
Size:	Maximum diameter: 2.1 in. (53 mm) 7.4 in. length (187.5 mm)
Net weight:	10.2 oz. (290 g)
Shipping weight:	2.2 lb. (970 g)
Accessories included:	SA 61 stand adapter, hard-shell case
Optional accessories:	W 23 foam windscreen B15 battery adaptor

Features

- **Broad high frequency rise (4 dB) delivers added presence and detail**
- **Supercardioid pickup pattern for excellent rejection of feedback**
- **Switchable low-frequency cut**
- **Internal elastic suspension for reduced handling noise**
- **Lightweight die-cast metal body with soft-touch dark matte finish**
- **Dent-resistant grille assembly with integral foam pop filter**
- **Universal phantom powering**
- **Shipped in metal hard-shell road case with SA-61 unbreakable stand adaptor**

Annexe 5

D-Link
Building Networks for People

24/48-Port Ethernet Switch

Web Smart Switch

- Choices of 24, 48 Ethernet Ports
- 4 Gigabit Uplinks
- Versatile SmartConsole Web-Based Management
- VLAN Traffic Segmentation & Priority Queue QoS
- Network Access Security
- Built-in SNMP MIB-II

FEATURES

- **Choices of 24 and 48 10/100BASE-TX Ethernet Ports**
- **2 10/100/1000T Gigabit Ports**
- **2 Combo 10/100/1000T/SFP Gigabit Ports**
- **Up to 6 Port Trunks for Server Connection/Switch Cascading**
- **802.1Q VLAN Tagging for Traffic Segmentation**
- **Up to 4 802.1p Priority Queues for QoS**
- **Access Security With MAC Address Filters & 802.1x Port-Based Authentication**
- **Broadcast Storm Control for Bandwidth Management**
- **Safeguard Engine Feature for Guaranteed Switch Performance**
- **Versatile SmartConsole Web-Based Management**
- **SNMP Management Support**
- **19-Inch Standard Rack-Mount Size**

smart™

D-Link's next generation Web Smart Ethernet switch series blends plug-&-play simplicity with exceptional performance and reliability to create a cost-effective solution for bandwidth-starved workgroups and departments. This series provides a solution for the small and medium-sized business (SMB) with different network size requirements. Each switch supports Gigabit uplink to servers, storage, or other switching devices. This series delivers superior performance with exceptional value, and an advanced feature set sufficient to monitor and secure a SMB network efficiently using simple web-based management.

Choices of 24 and 48 ports. Two port densities are available for selection: 24 Ethernet ports and 48 Ethernet ports. Supporting auto-detection of MDI/MDIX, these switches bring inexpensive and easy Ethernet connection to the desktops. Each switch provides 4 Gigabit uplinks connection to a Gigabit backbone or servers.

Extensive Layer 2 Features. Implemented as complete L2 devices, these switches include functions such as IGMP snooping, port mirroring, Spanning Tree, port trunks and 802.3x Flow Control to enhance performance and network resiliency.

Traffic Segmentation and QoS. The switches support 802.1Q VLAN Tagging for traffic segmentation by groups to enhance network security and performance. They also support 802.1p Priority Queues, enabling users to run bandwidth-sensitive applications such as streaming multimedia and VoIP on the network. These functions allow the switches to work seamlessly with managed VLAN and 802.1p traffic on the network.

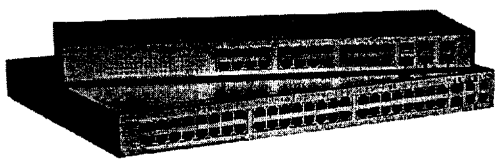
Network Security. The switches provide MAC address filters screen access to the network. They support 802.1x port-based authentication, allowing the network to be configured with external RADIUS servers. Additional features like D-Link Safeguard Engine protect the switches against traffic flooding caused by virus/worm outbreaks.

Web Smart Management. With full color graphic pictures, diagrams, and easy to understand navigation menus, the switches' SmartConsole Utility is as easy to use as surfing the net. It includes useful features like auto-discovery for automatic display of all Web Smart switches running on the network, Monitor List for status and trap messages and Trap View for system events, device boot up, abnormal data transfer errors and illegal login.

SNMP Management. With built-in SNMP-based MIBs, the switches can also be integrated in an SNMP-based network and polled to provide valuable information about the status of the unit and send traps on abnormal events.

DES-1228 24-Port Switch

DES-1252 48-Port Switch



DES-1228/1252

Annexe 6

Creating an IFB (Interrupt Foldback Bus)

An IFB is a special use of an output port. The person connected to the output port usually hears a program source connected to some input port of the intercom system. This program source is then interrupted when a keypanel operator presses a key to talk to the person. The following procedure lets you define which output port you want to use for IFB output, and which port will be used for program input. You can also assign a meaningful name for the IFB. Once you have configured an IFB, you can assign it to any keypanel key as described starting on page 2-15.

To set up an IFB:

1. From the menu bar, select the System menu, then select "IFB Buses". The IFB editing screen should appear (Figure 2-5). Interrupt Foldback Bus Assignments are shown in the table at the left of the screen. An Inputs/Outputs pick list is shown at the right. The default Alpha names for IFBs are IF01, IF02 etc. As you configure IFBs you will also change the Alphas to more meaningful names for your intercom system. The number of available IFBs will vary depending on the size of your intercom system.

2. Name the IFB: Use the UP/DOWN cursor keys to select an IFB Alpha (or click on the desired Alpha with a mouse). Then, enter a four-digit name for the IFB. (The IFB Alpha may be the same as the output port Alpha of the person that will be interrupted during IFB operation. Just remember that assigning the port to a talk key will have a different effect than assigning the IFB to a talk key.) When you move the cursor to a different position, you will notice that the Alias changes to the same name as the Alpha*. If your intercom system is interconnected (trunked) with another intercom system you may enter a different Alias name that will be meaningful to personnel in the other intercom system.

Note When you make any change to an IFB and move to a new position, you will notice that a check mark √ is inserted in the "Chg" column. This check mark is a change flag. (See the change flag description on page 1-4 for further details.)

3. Specify the output port of the person to be interrupted during IFB: Move to the "IFB Out" column. Type the Alpha of the output port, or select it from the pick list at the right side of the screen. In the example in Figure 2-5, port N033 has been entered as the IFB output for IF01.

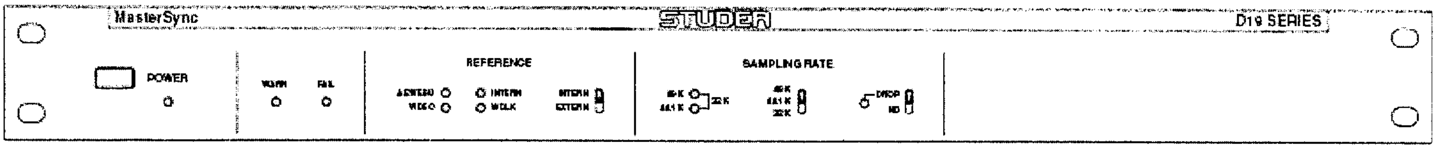
Note To use the pick list to insert ports into the Interrupt Foldback Bus Assignments table: Position the cursor in the table at the point where you want to make the insertion. Then, press the TAB key to move the cursor into the Inputs/Outputs pick list (or move the mouse cursor over the pick list). Select a port from the pick list and press ENTER (or click on a port with the left mouse button). The selected port should appear in the Interrupt Foldback Bus Assignments table.

* Except when an Alias has already been entered that is different from the Alpha. In this case, the Alias will not change whenever the Alpha is changed.

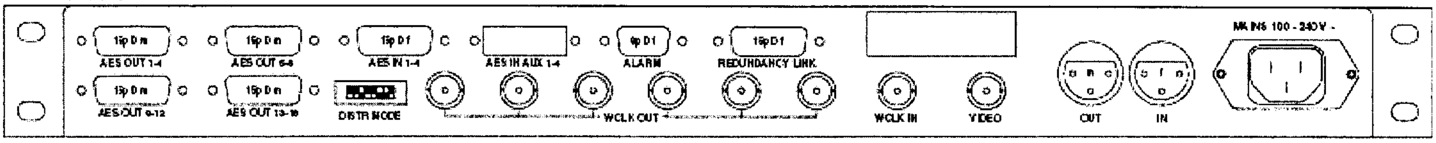
Annexe 7

13.9 Studer D19 MasterSync Generator

Front view:



Rear view:



The Studer D19 MasterSync generator/distributor is a 19"/1U unit. The generator part can be slaved to an external video clock, word clock, or AES/EBU input. Should the external reference signal fail, the generator automatically switches over to an internal reference with 1 ppm accuracy.
The unit distributes one word clock signal to 6 outputs, and the AES/EBU input (also AES/EBU frame clock signals) to typically 16 outputs each. The AES/EBU distribution can be configured as 1 × 16, 2 × 8, or 4 × 4 (or 1 × 8 and 2 × 4 simultaneously) distribution by means of a DIP switch on the back panel.
The generator can be set to 32 kHz, 44.1 kHz, 44.056 kHz, 48 kHz, and 47.952 kHz. When synchronizing to an external video reference, the generator rate can be set by means of switches.

One MasterSync generator is supplied with the D950 system as standard. Two generator/distributor units can optionally be linked together by means of a redundancy cable. In this case, one unit takes over the supply, the AES/EBU reference, and the word clock reference of both units, should the power supply of the other unit fail. Both units are always synchronized in normal operation, so that no phase shift can occur if one unit fails.

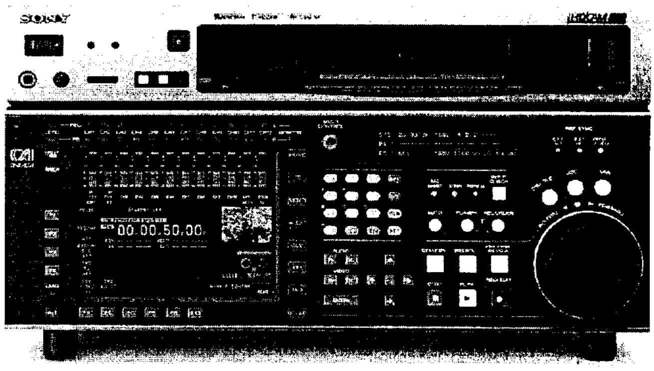
Typical power requirements:
100...240 V, 50/60 Hz, 50 W

Annexe 8

SRW-5000/5500 SPECIFICATIONS

	SRW-5000	SRW-5500
General		
Power requirements	100 to 240 V AC (±10%, 50/60 Hz)	
Power consumption	260 W (without options)/320 W (with all option boards installed)	
Operating temperature	+5 °C to +40 °C (+41 °F to +104 °F)	
Storage temperature	-20 °C to +60 °C (-4 °F to +140 °F)	
Operating humidity	25% to 80% (relative humidity)	
Mass (approx.)	30 kg (66 lb 2 oz)	
Dimensions (W x H x D excluding protrusions)	427 x 218 x 544 mm (16 ⅞ x 8 ⅝ x 21 ⅜ inches)	
Tape speed	HDCAM SR: 94.2 mm/s (24 Hz) HDCAM: 77.4 mm/s (24 Hz) Digital Betacam: 96.7 mm/s	
HDCAM SR/HDCAM* recording/ Playback time	155 min with BCT-124SR cassette (24 Hz) with BCT-124SRL or BCT-124HDL tape	
Digital Betacam playback time	124 minutes with BCT-D124L tape	
Fast-forward/rewind time	Approx. 4 min with BCT-124SR cassette	
Search speed range — Shuttle mode	HDCAM SR: Still to ±50 times normal playback speed (24 Hz) HDCAM: Still to ±58 times normal playback speed (25 Hz) Digital Betacam: Still to ±50 times normal playback speed	
Search speed range — Variable mode	HDCAM SR: -1 to 2 times normal playback speed HDCAM: -1 to 2 times normal playback speed Digital Betacam: -1 to 3 times normal playback speed	
Search speed range — Jog Mode	HDCAM SR: Still to ±2 times normal playback speed HDCAM: Still to ±3 times normal playback speed Digital Betacam: Still to ±3 times normal playback speed	
Dynamic Tracking Range	-1 to +2 times normal playback speed	
Servo-lock time	1.0 sec or less (from standby on)	
Load/unload time	7.0 sec or less	
Input/Output		
HD-SDI input A	BNC (1+ 1 for monitoring loop-through), Serial digital (1.485 Gb/s), SMPTE 292M/BTA S-004/ITU-R.BT 709	
HD-SDI input B (optional HKSR-5003 required)	BNC (1+ 1 for monitoring loop-through), Serial digital (1.485 Gb/s), SMPTE 292M/BTA S-004/ITU-R.BT 709	
HD/SD reference video input 1	BNC (1 + 1 for loop-through), Tri Level sync, 0.6 Vp-p, 75 Ω , sync negative or Black Burst, 0.286 Vp-p, 75 Ω , sync negative	
HD/SD reference video input 2 (optional HKSR-5001 required)	BNC (1 + 1 for loop-through), Tri Level sync, 0.6 Vp-p, 75 Ω , sync negative or Black Burst, 0.286 Vp-p, 75 Ω , sync negative	
Digital-audio input (CH1/2, CH3/4, CH5/6, CH7/8, CH9/10, CH11/12)	BNC (x6, AES/EBU), unbalanced	
Analogue audio input (Cue)	—	XLR-3-pin, female x1
Time-code input	XLR-3-pin type, (female x1), 0.5 to 18 Vp-p, 10 kΩ , balanced	
HD-SDI output	BNC (2 + 1, with character out), Serial digital (1.485 Gb/s), SMPTE 292M/BTA S004/ITU-R.BT 709	
Format-converter output (optional HKSR-5001 required)	BNC (x2), with character out	
SD-SDI output	BNC (2 + 1 with character out), D1 serial digital (270 Mb/s), SMPTE 259M	
Analogue down-converted output	Composite: BNC (x1 with character out) 1.0 Vp-p, 75 Ω , sync negative) SD sync: BNC (x1, Black Burst, 0.286 Vp-p, 75 Ω , sync negative) output	
Analogue reference output	1125 Sync: BNC (x2), Tri Level sync, 0.6 Vp-p, 75 Ω , sync negative	
Digital-audio output (CH1/2 CH3/4 CH5/6 CH7/8 CH9/10 CH11/12)	BNC (x6), AES/EBU, unbalanced	
Analogue-audio output (CH1/2/3/4/Cue**)	XLR-3-pin type, (male x5), +4 dBm, (with a 600 Ω load), low impedance, balanced	
Monitor output (L/R)	XLR-3-pin type, (male x2), +4 dBm, (with a 600 Ω load), low impedance, balanced	
Time-code output	XLR-3-pin type, (male x1), 2.2 Vp-p low impedance, balanced	
Phones	JM-60 stereo phone jack, -∞ to 12 dBu (with an 8 Ω load), unbalanced	
Remote 1 input	D-sub 9-pin, (female), Sony 9-pin remote interface	
Remote 1 input/output	D-sub 9-pin, (female), Sony 9-pin remote interface	
Video control	D-sub 9-pin, (female), (for optional HKDV-900)	
Parallel remote	D-sub 50-pin, (female)	
Ethernet	10Base-T modular jack	
Digital Video Performance		
Sampling frequency	HDCAM SR: Y: 74.25 MHz, Cb/Cr: 37.125 MHz, G/B/R: 74.25 MHz HDCAM*: Y: 74.25 MHz, Cb/Cr: 37.125 MHz	
Quantisation	10 bits/sample	
Compression	HDCAM SR: MPEG-4 Studio Profile HDCAM*: Coefficient Recording System	
Channel coding	S-NRZ	
Error correction	Reed-Solomon code	
Error concealment	Adaptive three-dimensional	
Analogue Composite Output Performance		
Bandwidth	Y: 0 to 5.75 MHz +0.5 dB/-3.0 dB	
S/N ratio	56 dB or more	
Y/C delay	15 ns or less	
K Factor (2T Pulse)	1% or less	
Output SCH phase	Based upon RS-170A/CCIR R.624-3	
Digital Audio Performance		
Sampling frequency	48 kHz (synchronised with video)	
Quantisation	HDCAM SR: 24 bits/sample HDCAM*: 20 bits/sample	
Wow & flutter	Below measurable level	
Headroom	20/18/16/12 dB selectable	
Analogue Audio Output Performance		
D/A quantisation	24 bits/sample	
Frequency response	20 Hz to 20 kHz, +0.5 dB/-1.0 dB (0 dB at 1 kHz)	
Dynamic range	More than 100 dB (At 1dB at 1 kHz)	
Distortion	Less than 0.05% (At 1 kHz, reference level)	
Crosstalk	Less than -80 dB (At 1 kHz, between any two channels)	
De-emphasis	T1 = 50 µs, T2 = 15 µs (auto on/off)	
Supplied Accessories		
	Operation manual, installation manual	

* The SRW-5000 does not support HDCAM recording.
** HDCAM and Digital Betacam playback only.



Annexe 9

To facilitate the conversion of TV production and broadcast facilities to multichannel audio, Dolby Laboratories has developed a new, professional digital audio coding system, Dolby E. With Dolby E, up to eight channels of high-quality audio plus Dolby Digital metadata (see sidebar) can be distributed via an AES3 pair, or recorded onto two audio tracks of a digital VTR.

Instead of having to replace their audio equipment and routing systems, many facilities can convert to multichannel audio simply by adding Dolby E codecs to their existing two-channel AES3 distribution systems. The result is efficient, cost-effective post-production and distribution of multichannel programs prior to final Dolby Digital (AC-3) encoding and transmission.

Dolby E is a professional system for use within the broadcast and post-production infrastructure. Audio never reaches the consumer in Dolby E form; it is encoded with Dolby Digital just prior to final transmission. To help differentiate their functions, Dolby E is referred to as a distribution coding system, and Dolby Digital as an emission coding system.

Some industry experts predict that thrilling 5.1-channel Dolby Digital sound could be even more vital to DTV's success than its improved picture. Multichannel audio cannot reach the viewer, however, if it never gets to the transmitter. Dolby E will help ensure that it does.

Dolby E at a Glance

- Simple, cost-effective conversion of two-channel broadcast and post-production facilities to multichannel audio.
- Distributes eight channels of high-quality audio and Dolby Digital metadata via AES3 or digital VTR audio track pair.
- Up to ten encode/decode cycles without degradation.
- Glitch-free audio editing synchronous with video within digital domain.
- Compatible with international video standards.

Why another audio coding system?

With its combination of quality, multichannel capability, and very low data rate, Dolby Digital coding is ideal for transmitting multichannel audio to the DTV viewer at home. It has been adopted as the standard audio coding in ATSC countries and is gaining favor as an alternative audio format for DVB countries.

Dolby Digital audio coding is not appropriate, however, for distributing multichannel audio within professional post-production and broadcast environments. Because it is optimized for maximum quality at low bit rates, it is limited to a single cycle of encoding (transmission) and decoding (reception). Also, because its frames do not match video frames, Dolby Digital audio is not optimized for editing when changes to the picture are needed.

Dolby E, on the other hand, has been developed specifically for distribution, rather than emission, for applications such as sending a program to a local station for commercial insertion, routing it within the same studio for voice-over editing, or sending it via satellite to another broadcast facility. As a result of its sophisticated algorithm and higher data rate, Dolby E programs can withstand up to ten tandem encode/decode cycles without audible degradation.

With Dolby E, audio frames match video frames, assuring that audio-follow-video edits are free of mutes, glitches, or other aberrations. It also makes it possible to switch, route, and perform assemble edits directly on the Dolby E bitstream without decoding and re-encoding.

Using Dolby E

Dolby E encodes up to eight audio channels plus metadata into a two-channel bitstream with a standard data rate of 1.92 Mbits/sec (20-bit audio at 48 kHz). With multichannel programming, a "5.1+2" configuration is typically used, with six of the eight channels carrying a 5.1 mix and the other two an Lt/Rt (matrix surround-encoded) or stereo two-channel mix. The system can also be used to carry a 5.1 mix plus two mono tracks (5.1+1+1), three stereo mixes (3x2), six mono channels (6x1), and so on.

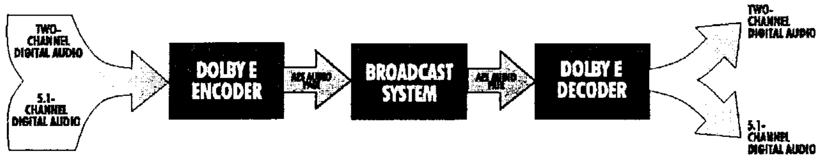


Figure 1: Dolby E in the distribution chain

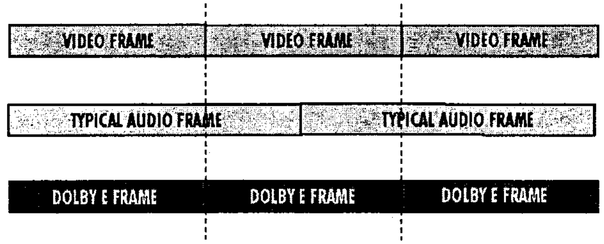


Figure 2: Dolby E frames match video frames for smooth editing

Dolby E adds one frame of delay during encoding and one frame during decoding, and requires a "color-black" reference signal to keep audio and video frame rates locked.

Dolby E has been designed to work with most international video standards. Initially, it supports frame rates of 29.97 fps, 20-bit output audio words, and a 48 kHz sample rate, while in the future it will also support 25 fps, and 16-bit and 24-bit audio words.

Dolby E in post-production

As more and more broadcast facilities equip with Dolby E, post-production facilities will be increasingly called upon to deliver mixes in the two-track, Dolby E encoded format. While this alone is sufficient reason to use Dolby E, its advantages are as useful in post-production as in program distribution. The ability to fit eight channels plus Dolby Digital metadata into a two-channel architecture, to accomplish audio-follow-video editing, and to maintain audio quality can all help facilitate the post-production of audio destined for DTV transmission.

Most important of all, the use of Dolby E in post-production lets mixers include metadata parameters along with the mix that are carried down the line through the distribution chain to the Dolby Digital encoder. As a result, the mix reaches viewers at home exactly as originally produced, so they hear precisely what the program's creators intended.

Dolby Digital encoding: the final step

Audio for DTV programs should be maintained in the robust Dolby E format right up to final master control, and only then re-encoded as a Dolby Digital data stream for transmission. Doing so will ensure the highest possible audio quality for the viewer at home, while at the same time simplifying the distribution process.

What is metadata?

The Dolby Digital emission coding system was designed not only to be highly efficient, but also to satisfy all viewers, from those with mono TV sets in noisy environments to those with elaborate multichannel home theater systems capable of a wide dynamic range.

To this end, the program producer can incorporate within the Dolby Digital bitstream auxiliary information called metadata (i.e., data about the data) to control aspects of the decoding and reproduction of the audio at the viewer's location. Listeners can then apply, partially apply, or ignore these parameters as appropriate to their equipment and preferences.

One metadata parameter can, for example, signify the program's number and type of channels (audio coding modes). Another, called *dynrange*, can be used to compress the audio's dynamic range by a predetermined amount when appropriate (such as late at night), yet allow listeners to opt for full dynamic range when they prefer. And *dialnorm* is used to automatically adjust the consumer decoder's output level to produce consistent playback loudness on all programs, including commercials.

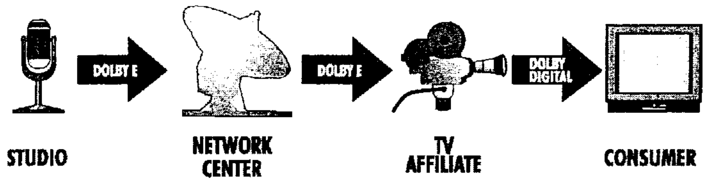


Figure 3: Dolby E and Dolby Digital distribute DTV productions to consumers