Annexe 4

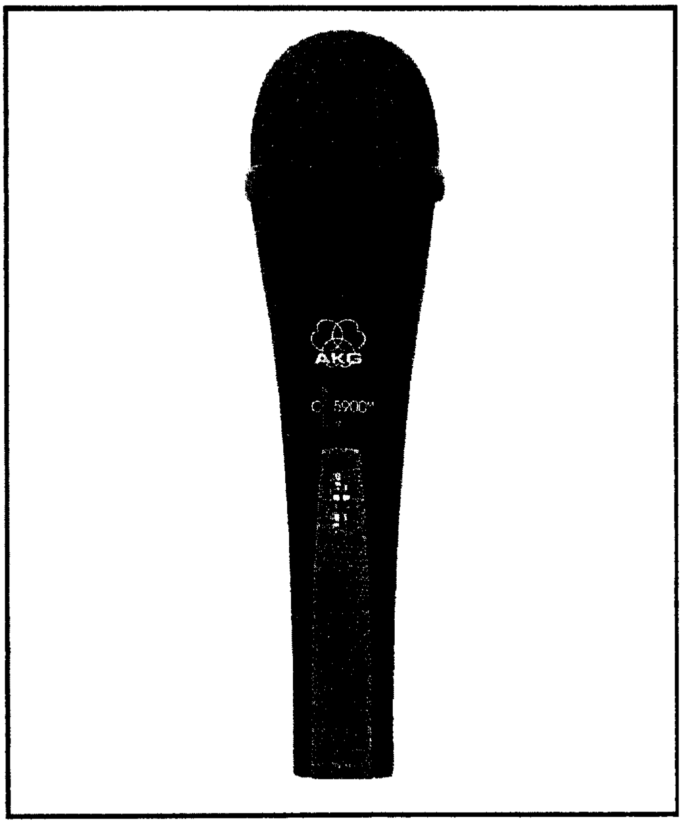
AKG
H A Harman International Company

# 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)<br>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<br>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.

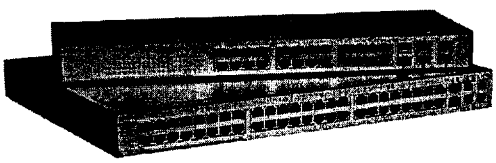
*Network Security.* The switches provide MAC address filters screen access to the network. They support 802.1x port-based authentication, allowing the network to be configured with external RADIUS servers. Additional features like D-Link Safeguard Engine protect the switches against traffic flooding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



**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.

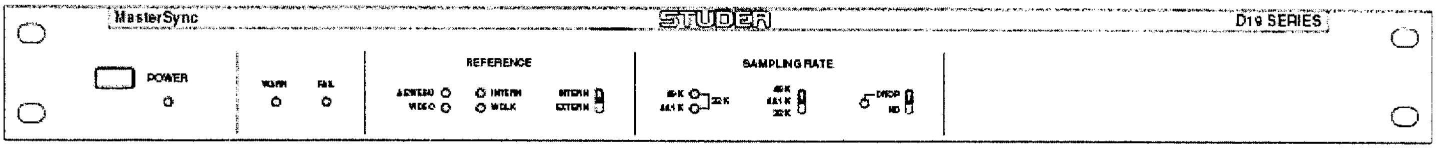
**Note** To use the pick list to insert ports into the Interrupt Foldback Bus Assignments table: Position the cursor in the table at the point where you want to make the insertion. Then, press the TAB key to move the cursor into the Inputs/Outputs pick list (or move the mouse cursor over the pick list). 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.

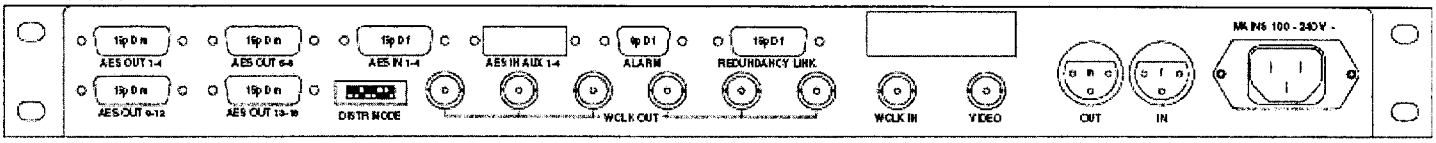
# Annexe 7

## 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.

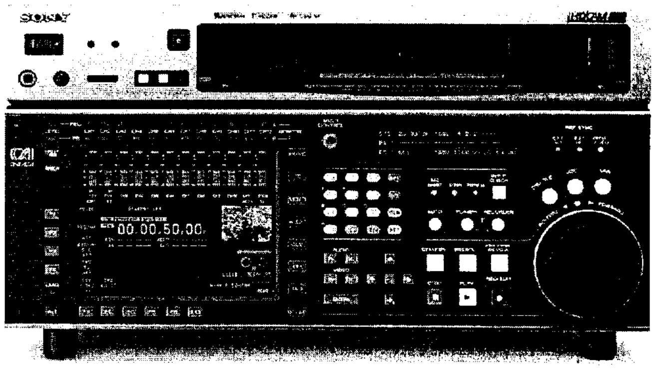
**Typical power requirements:**
100...240 V, 50/60 Hz, 50 W

## 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)<br>HDCAM: 77.4 mm/s (24 Hz)<br>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)<br>HDCAM: Still to ±58 times normal playback speed (25 Hz)<br>Digital Betacam: Still to ±50 times normal playback speed | |
| Search speed range – Variable mode | HDCAM SR: -1 to 2 times normal playback speed<br>HDCAM: -1 to 2 times normal playback speed<br>Digital Betacam: -1 to 3 times normal playback speed | |
| Search speed range – Jog Mode | HDCAM SR: Still to ±2 times normal playback speed<br>HDCAM: Still to ±3 times normal playback speed<br>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)<br>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<br>HDCAM*: Y: 74.25 MHz, Cb/Cr: 37.125 MHz | |
| Quantisation | 10 bits/sample | |
| Compression | HDCAM SR: MPEG-4 Studio Profile<br>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<br>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.

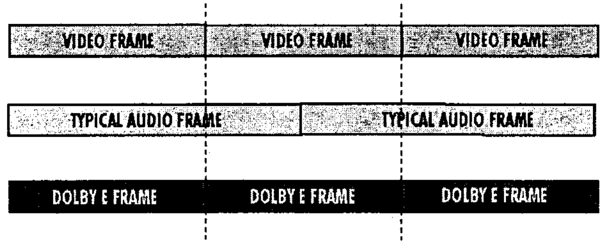



Figure 2: Dolby E frames match video frames for smooth editing

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.

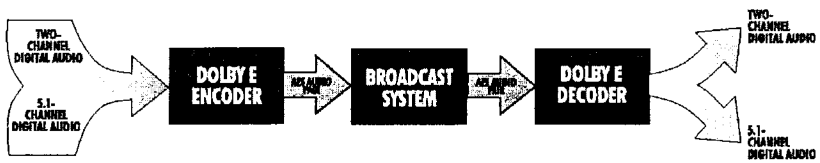



Figure 1: Dolby E in the distribution chain

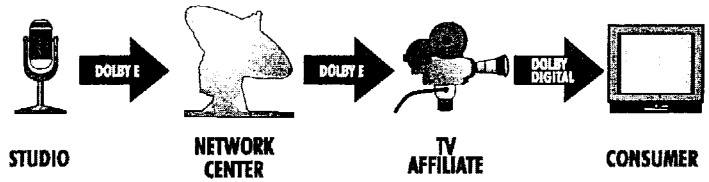



Figure 3: Dolby E and Dolby Digital distribute DTV productions to consumers