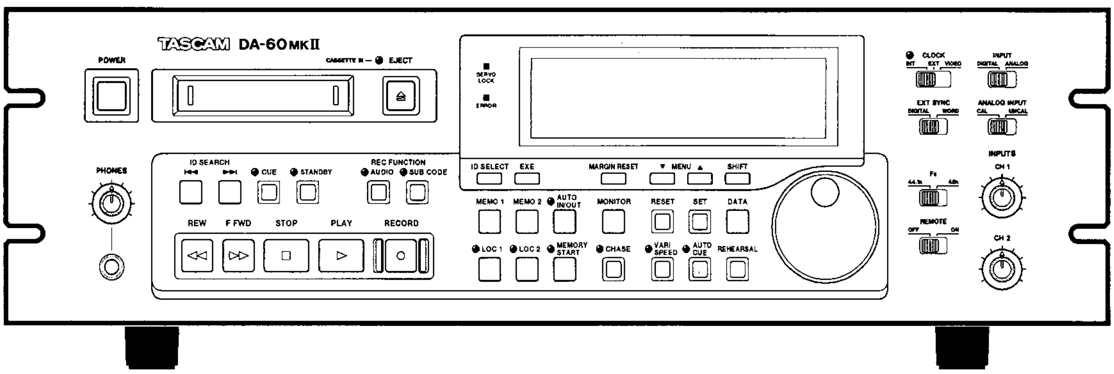
TASCAM

TEAC Professional Division

DA-60MKII

Digital Audio Tape Deck



OWNER'S MANUAL

D00212900A

Important Safety Precautions

CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE COVER (OR BACK). NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

The lightning flash with arrowhead symbol, within equilateral triangle, is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to person.

The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

This appliance has a serial number located on the rear panel. Please record the model number and serial number and retain them for your records.
Model number ____________
Serial number ____________

WARNING: TO PRIVENT FIRE OR SHOCK HAZARD, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.

For U.S.A

TO THE USER

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications.
Operation of this equipment in a residental area is likely to cause harmful interference in witch case the user will be required to correct the interference at his own expense.

CAUTION
Changes or modifications to this equipment not expressly approved by TEAC CORPORATION for compliance could void the user's authority to operate this equipment.

For the consumers in Europe

WARNING
This is a Class A product. In a domestic environment, this product may cause radio interference in which case the user may be required to take adequate measures.

Pour les utilisateurs en Europe

AVERTISSEMENT
Il s'agit d'un produit de Classe A. Dans un environnement domestique, cet appareil peut provoquer des interférences radio, dans ce cas l'utilisateur peut être amené à prenre des mesures appropriées.

Für Kunden in Europa

Warnung
Dies is eine Einrichtung, welche die Funk-Entstörung nach Klasse A besitzt. Diese Einrichtung kann im Wohnbereich Funkstörungen versursachen ; in dieasem Fall kann vom Betrieber verlang werden, angemessene Maßnahmen durchzuführen und dafür aufzukommen.

SAFETY INSTRUCTIONS

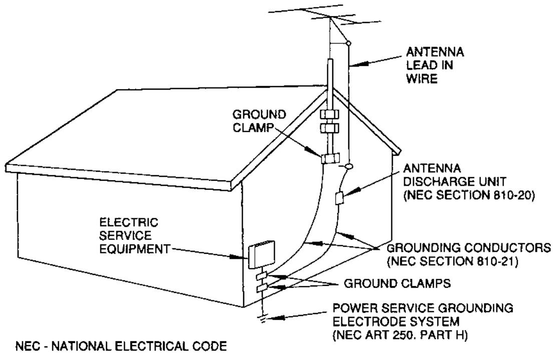
CAUTION:
- **Read all of these Instructions.**
- **Save these Instructions for later use.**
- **Follow all Warnings and Instructions marked on the audio equipment.**

1) Read instructions — All the safety and operating instructions should be read before the product is operated.
2) Retain instructions — The safety and operating instructions should be retained for future reference.
3) Heed Warnings — All warnings on the product and in the operating instructions should be adhered to.
4) Follow instructions — All operating and use instructions shoud be followed.
5) Cleaning — Unplug this product from the wall outlet before cleaning. Do not use liquid cleaners or aerosol cleaners. Use a damp cloth for cleaning.
6) Attachments — Do not use attachments not recommended by the product manufacturer as they may cause hazards.
7) Water and Moisture — Do not use this product near water - for example, near a bath tub, wash bowl, kitchen sink, or laundry tub; in a wet basement; or near a swimming pool; and the like.
8) Accessories — Do not place this product on an unstable cart, stand, tripod, bracket, or table. The product may fall, causing serious injury to a child or adult, and serious damage to the product. Use only with a cart, stand, tripod, bracket, or table recommended by the manufacturer, or sold with the product. Any mounting of the product should follow the manufacturer's instructions, and should use a mounting accessory recommended by the manufacturer.
9) A product and cart combination should be moved with care. Quick stops, excessive force, and uneven surfaces may cause the product and cart combination to overturn.

10) Ventilation — Slots and openings in the cabinet are provided for ventilation and to ensure reliable operation of the product and to protect it from overheating, and these openings must not be blocked or covered. The openings should never be blocked by placing the product on a bed, sofa, rug, or other similar surface. This product should not be placed in a built-in installation such as a bookcase or rack unless proper ventilation is provided or the manufacturer's instructions have been adhered to.
11) Power Sources — This product should be operated only from the type of power source indicated on the marking label. If you are not sure of the type of power supply to your home, consult your product dealer or local power company. For products intended to operate from battery power, or other sources, refer to the operating instructions.
12) Grounding or Polarization — This procuct may be equipped with a polarized alternating-current line plug (a plug having one blade wider than the other). This plug will fit into the power outlet only one way. This is a safety feature. If you are unable to insert the plug fully into the outlet, try reversing the plug. If the plug should still fail to fit, contact your electrician to replace your obsolete outlet. Do not defeat the safety purpose of the polarized plug.
13) Power-Cord Protection — Power-supply cords shoud be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords at plugs, convenience receptacles, and the point where they exit from the product.
14) Outdoor Antenna Grounding — If an outside antenna or cable system is connected to the product, be sure the antenna or cable system is grounded so as to provide some protection against voltage surges and built-up static charges. Article 810 of the National Electrical Code, ANSI/NFPA 70, provides information with regard to proper grounding of the mast and supporting structure, grounding of the lead-in wire to an antenna discharge unit, size of grounding conductors, location of antenna-discharge unit, connection to grounding electrodes, and requirements for the grounding electrode.

"Note to CATV system installer:
This reminder is provided to call the CATV system installer's attention to Section 820-40 of the NEC which provides guidelines for proper grounding and, in particular, specifies that the cable ground shall be connected to the grounding system of the building, as close to the point of cable entry as practical.

Example of Antenna Grounding as per National Electrical Code, ANSI/NFPA 70



15) Lightning — For added protection for this product during a lightning storm, or when it is left unattended and unused for long periods of time, unplug it from the wall outlet and disconnect the antenna or cable system. This will prevent damage to the product due to lightning and power-line surges.
16) Power Lines — An outside antenna system should not be located in the vicinity of overhead power lines or other electric light or power circuits, or where it can fall into such power lines or circuits. When installing an outside antenna system, extreme care should be taken to keep from touching such power lines or circuits as contact with them might be fatal.
17) Overloading — Do not overload wall outlets, extension cords, or integral convenience receptacles as this can result in risk of fire or electric shock.
18) Object and Liquid Entry — Never push objects of any kind into this product through openings as they may touch dangerous voltage points or short-out parts that could result in a fire or electric shock. Never spill liquid of any kind on the product.
19) Servicing — Do not attempt to service this product yourself as opening or removing covers may expose you to dangerous voltage or other hazards. Refer all servicing to qualified service personnel.
20) Damage Requiring Service — Unplug this product from the wall outlet and refer servicing to qualified service personnel under the following conditions:
a) when the power-supply cord or plug is damaged.
b) if liquid has been spilled, or objects have fallen into the product.
c) if the product has been exposed to rain or water.
d) if the product does not operate normally by following the operating instructions. Adjust only those controls that are covered by the operating instructions as an improper adjustment of other controls may result in damage and will often require extensive work by a qualified technician to restore the product to its normal operation.
e) if the product has been dropped or damaged in any way.
f) when the product exhibits a distinct change in performance - this indicates a need for service.
21) Replacement Parts — When replacement parts are required, be sure the service technician has used replacement parts specified by the manufacturer or have the same characteristics as the original part. Unauthorized substitutions may result in fire, electric shock, or other hazards.
22) Safety Check — Upon completion of any service or repairs to this product, ask the service technician to perform safety checks to determine that the product is in proper operating condition.
23) Wall or Ceiling Mouting — The product shoud be mounted to a wall or ceiling only as recommended by the manufacturer.
24) Heat — The product should be situated away from heat sources such as radiators, heat registers, stoves, or other products (including amplifiers) that produce heat.

SECTION 1 : INTRODUCTION

The DA-60 MKII was designed for professional use and its features include the following :

- ○ 4-head, 4-DD mechanism
- ○ Punch-in/out (with digital crossfade)
- ○ Variable speed playback
- ○ Auto Cue/Memory Start ensuring instant play start from the exact point
- ○ Autolocation
- ○ Fade in and out of play or record
- ○ Digital audio interface conforming to AES/EBU standards
- ○ External sync (referenced to Word Sync signal)
- ○ Parallel data port and RS-422 serial data port for interface to external controllers
- ○ Record and read of SMPTE/EBU/FILM timecodes, following IEC standards
- ○ Timecode generator integrated
- ○ Timecode-controlled synchronization
- ○ Slaving to an incoming composite video or a film sync signal

Using this manual

Before actually using the DA-60 MKII, please read this manual thoroughly at least once, so you will know where to return to when you need answers. Even though a quick glance will get you going, careful study will ensure that misunderstandings won't slow you down.

Use of capital letters : In general, we use all upper case type to designate a particular switch, control or connector label.

Installation Site

The DA-60 MKII may be used in most area, but to maintain top performance and prolong operating life, observe the following environmental limitations :

1) Nominal temperature should be 5 to 35 degrees Centigrade (41 to 95 degrees Fahrenheit).

2) Relative humidity should be 30 to 90% (non-condensing).

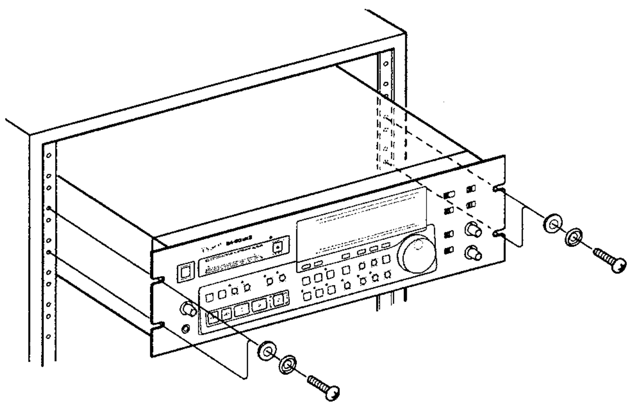
3) Strong magnetic fields should not exist nearby.

Beware of Condensation

When the DA-60 MKII is moved from a cold to a warm place or used after sudden temperature change, there is the danger of condensation ; water vapor in the air could condense on the internal mechanism, making correct operation impossible. To prevent this, or if this occurs, leave the DA-60 MKII for 1 or 2 hours with power turned on, then turn off power and switch on again.

Mounting in a 19" Rack

○ Be sure to leave a 1 U or more space on top of the unit.



Care of the DA-60 MKII

- Be careful not to drop your deck or don't subject it to severe impact, especially during recording.
- When cleaning the exterior of the deck, use a soft cloth. If necessary, moisten a soft cloth with mild solution of detergent and water. Do not use any type of solvents such as alcohol or benzine.

Table of Contents

Important (for U.K. Customers)

DO NOT cut off the mains plug from this equipment. If the plug fitted is not suitable for the power points in your home or the cable is too short to reach a power point, then obtain an appropriate safety approved extension lead or consult your dealer.

If nonetheless the mains plug is cut off, remove the fuse and dispose of the plug immediately, to avoid a possible shock hazard by inadvertent connection to the mains supply.

If this product is not provided with a mains plug, or one has to be fitted, then follow the instructions given below:

IMPORTANT: The wires in this mains lead are coloured in accordance with the following code:

GREEN-AND-YELLOW	:	**EARTH**
BLUE	:	**NEUTRAL**
BROWN	:	**LIVE**

WARNING: This apparatus must be earthed.

As the colours of the wires in the mains lead of this apparatus may not correspond with the coloured markings identifying the terminals in your plug proceed as follows:

The wire which is coloured GREEN-and-YELLOW must be connected to the terminal in the plug which is marked by the letter E or by the safety earth symbol ⏚ or coloured GREEN or GREEN-and-YELLOW.

The wire which is coloured BLUE must be connected to the terminal which is marked with the letter N or coloured BLACK.

The wire which is coloured BROWN must be connected to the terminal which is marked with the letter L or coloured RED.

When replacing the fuse only a correctly rated approved type should be used and be sure to re-fit the fuse cover.

IF IN DOUBT — CONSULT A COMPETENT ELECTRICIAN.

Head Drum Replacement

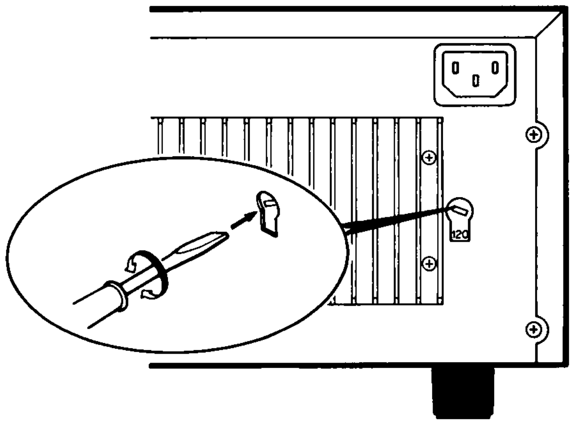
TASCAM recommends that you replace the head drum after every about 1,000 hours of operation for quality recording and playback. When 1,000 hours more or less are shown at a "2 Hour" menu, contact TASCAM or you nearest TASCAM dealer.

Voltage Conversion (General Export Models Only)

NOTE

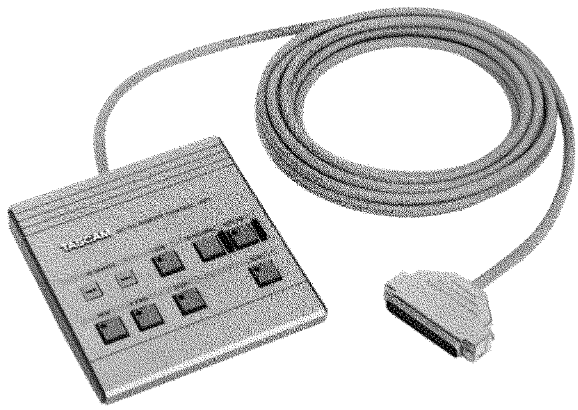
Voltage conversion is not possible on models sold in the U.S.A., Canada, U.K., Australia or Europe.

For general export models only, if the input voltage specified on the machine, power cord tag, or packing cartonm differs from the leine voltage at eh installation site, first make sure that AC power cord id disconnected, then locate the voltage slector on the rear panel (figure below), and turn the selector using an appropritate screwdriver until the required voltage appears.



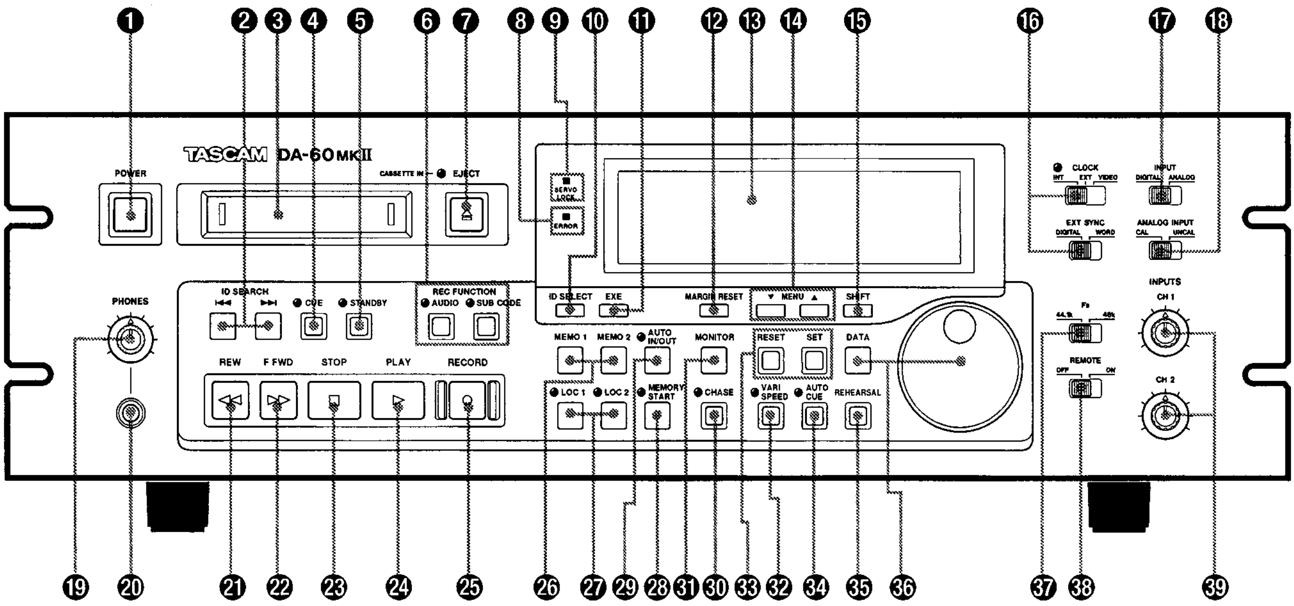
Optional Accessory

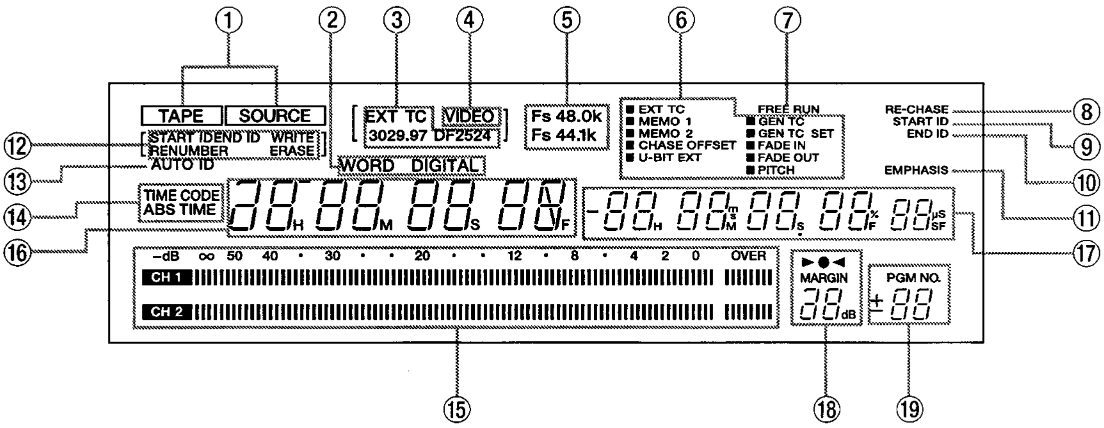
RC-D6 Remote Control Unit



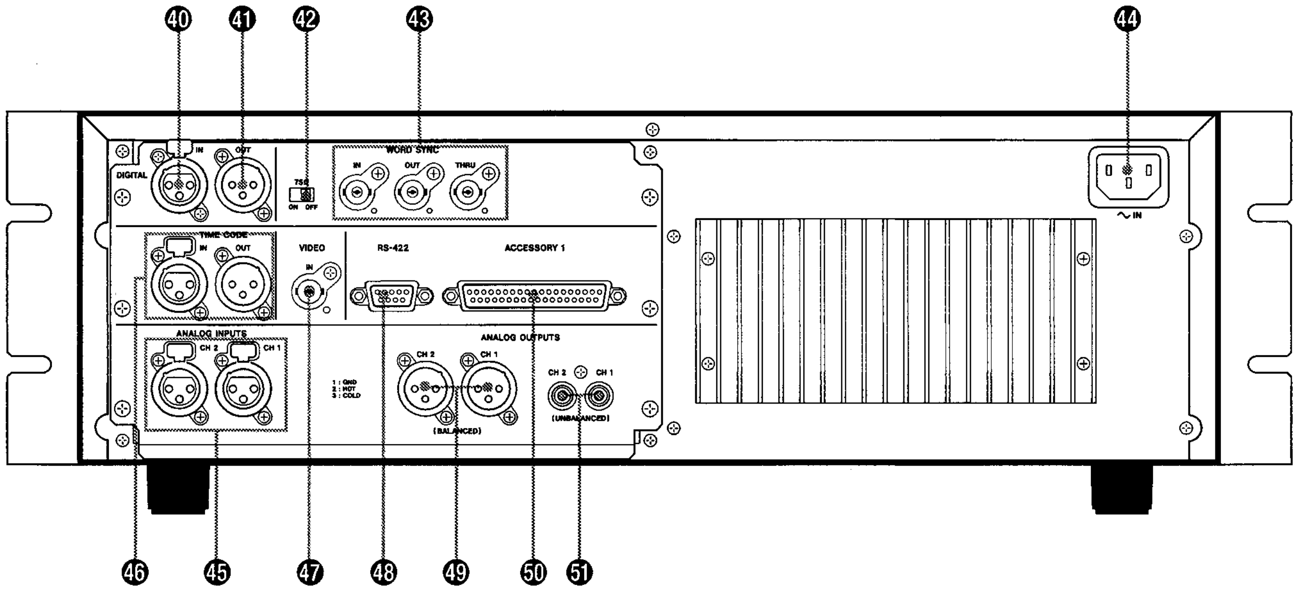
SECTION 2 : FEATURES AND CONTROLS

Front Panel





Rear Panel



Skimming through this section of the manual will allow you to get an overview of the DA-60 MKII. It is not necessary to memorize all of what is here nor to try to understand all details to get started.

2-1. Front Panel

❶ POWER switch

Controls the power to the DA-60 MKII. Turning off power does not reset your settings at menus. The on/off state of the MEMORY START (item 28) and AUTO CUE (item 34) functions is or is not saved, as selected at a menu.

❷ ID SEARCH keys

Can operate during play or stop, and have the tape locate to the next or the previous Start ID mark. The transport will go into Standby mode at the end of search operations.

❸ IDCassette loading slot

❹ CUE switch

When pressed, it causes the deck to be ready for review or cueing, waiting for you to press F.FWD or REW. When either of these is pressed the first time, the PLAY button will light, also the fast wind button pressed, and you can hear the tape at about 1 time normal play speed. A second press, causes the tape to play at about 3 times normal play speed. A third press, speeds up to about 9 times normal play speed.

Pressing PLAY during CUE mode slows the tape down to the normal play speed. Pressing CUE again disables the mode.

❺ STANDBY switch

STANDBY is a mode in which, if the tape is not running, the head drum is spinning. This ensures a tight start of play (or record).

Initiating PLAY, F.FWD or REW automatically causes the STANDBY LED to turn on, showing that the head drum will go into STANDBY mode when you press STOP.

To disable STANDBY mode, press the switch again.

☞ *If left in STANDBY mode for about 3 minutes, the deck will automatically exit the mode to avoid head wear.*

❻ REC FUNCTION switches

Offer three record/write modes, as discussed later (p. 5 • 1)

❼ EJECT Key

Used to remove the cassette tape.

❽ ERROR indicator

Lights during play when errors occur in the digital data at so excessive rates that they cannot be corrected and are submitted to interpolation to arrive at an approximation to the correct data.

❾ SERVO LOCK indicator

Lights when the tape is correctly running, or when the master and slave transports are locked in sync.

❿ ID SELECT switch

Each time this switch is pressed the following modes are selected in sequence, as shown in the display :

1. START ID WRITE (in ASSEMBLE or EDIT SUB mode)
2. START ID ERASE (in EDIT SUB mode)
3. END ID WRITE (in ASSEMBLE mode)
4. RENUMBER (in EDIT SUB mode)

Pressing the next EXE key executes selected operations.

⓫ EXE key

Used to execute operations selected by the ID SELECT switch.

⓬ MARGIN RESET key

Defeats a MARGIN (headroom available) indication so new readings can be taken.

⓭ Display window

Provides various information and messages, keeping you aware of what is currently taking place.

① **Monitor Source : TAPE** lights to show that the audio outputs and the meters are fed on tape signal. **SOURCE** lights when the monitor is switched to the input.

② **Sync Signal : WORD** will light when the deck is referenced to the Word Sync input. **DIGITAL** will light to show that the deck is referenced to the clock derived from the digital in.

③ **Time Code In : EXT TC** lights when time codes are being fed into the deck from the exterior. The type of time code selected at a "22 rEFtc" menu is also shown.

④ **Video/Film In : VIDEO** lights when the clock derived from the video in.

⑤ **Sampling Frequency :** Either indication lights when recording from analog input, as optionally selected ; or, when recording from digital input or during play, the sampling frequency at which the original recording was made will show.

⑥ **Menu Items and Variables :** Menus, and options available at them, are shown here. For more information, see Section 17, List of Menus.

⑦ **FREE RUN :** Blinks when the timecode generator is in the corresponding mode. See page 7 • 3 for an explanation.

⑧ **RE-CHASE :** Lights when this unit is used as a slave machine and is in the corresponding mode. For more information, see page 15 • 2.

⑨ **START ID :** Lights each time a Start ID mark is encountered.

⑩ **END ID :** Lights when an End ID mark is encountered.

⑪ **EMPHASIS** : Lights during both the pre-emphasis and the de-emphasis process.

⑫ **Subcode Edit Mode Indications :** The following are lit in sequence as you press the ID SELECT switch :

START ID WRITE : Shows that you can write Start ID marks manually (in ASSEMBLE or EDIT SUB mode).

START ID ERASE : Shows that you can erase Start ID marks (in EDIT SUB mode only).

END ID WRITE : Shows that you can mark the end of audio recordings with End ID (in ASSEMBLE mode only).

RENUMBER : You have to let this indication light when renumbering programs on tape (in EDIT SUB mode).

⑬ **AUTO ID :** Lights when recording of Start ID is automated.

⑭ **Time Indicators : TIME CODE** lights when professional DAT timecode is being converted to SMPTE/EBU timecode. When ABS time is being converted to SMPTE/EBU timecode, both the **TIME CODE** and the **ABS TIME** indicators are lit.

⑮ **Peak Level Meters :** If inputs overreach the meter scale, the red OVER indicator comes on. You can select a peak-hold mode at a menu, and also the release time of readings.

⑯ **Time Readout (left) :** Shows timecode numbers or menus. For details of the menus, see Section 17.

⑰ **Time Readout (right) :** Shows your settings at menus or locating time points.

⑱ **MARGIN indicator**
This is a digital peak-hold meter, showing the available headroom before digital saturation is reached and distortion occurs. It holds the highest reading since MARGIN RESET was last pressed (or since a new tape has been loaded). Readings range from 39 dB down to 0 dB (in 1 dB steps). If you have "1 Audio" (page 17 • 1) show on the left hand side of the display, the "●" indicator will light at the reference margin of "16 dB"; and at higher readings the "◀" indicator will light, and at lower readings the "▶" indicator will light.

⑲ **PGM NO. :** Shows (1) program numbers (if available) as the tape runs; (2) how many times you have pressed the ID SEARCH key to locate the tape to the Start ID mark of a program (p.10 • 1) ; and (3) frame numbers when trimming a MEMORY START point (p.11 • 1).

⓮ MENU key
Used to access menus. Depending on menus, you have to press SHIFT at the same time, as discussed later.

⓯ SHIFT key
Used together with the MENU key (or with the DATA dial) to access menus or to make your selections at them.

⓰ CLOCK and EXT SYNC select switches
Used to select a signal to which the DA-60 MKII is referenced for synchronization. See page 5 • 3 for details.

⓱ INPUT select switch
Used to select either the digital input or the analog input as the source of the DA-60 MKII.

⓲ ANALOG INPUT select switch
Determines whether the analog input passes through the level controls before reaching the tape and the output.

☞ *This switch has effect only when the INPUT source select switch is set to the right/ANALOG position, and the monitor is switched to the input, as confirmed by the "SOURCE" indication being lit in the display.*

CAL : The CH 1 and CH 2 INPUTS level controls are bypassed and the input level is passed to the output without receiving any level alteration. This CAL position is generally used for system calibration (level alignment). When a nominal level input is fed in, the level meter will read -16 dB. The nominal output level is +4 dBm at the balanced XLR-type connectors, and -10 dBV at the unbalanced RCA jacks.

UNCAL : The INPUTS level controls can operate.

⑲ PHONES control
Used to adjust the listening level in the headphones plugged into the jack just below.

⑳ PHONES jack
For connection of stereo headphones.

> *CAUTION*
>
> *Don't connect a 2-conductor mono plug to this jack which will short out one of the amplifiers feeding the headphones, causing it burn out.*

㉑ REW button
Winds the tape at high speed in reverse. If pressed when the CUE indicator lights, you can hear it play at high speeds. See also item 4.

㉒ F.FWD button
Similar to REW, but winds the tape in the forward direction.

㉓ STOP button
Disables the current transport mode and stops any tape motion. A STANDBY feature is then automatically activated and the head drum continues to spin for the resumption of play (or record) to be instantaneous.

㉔ PLAY button
Enables play mode. If REC FUNCTION is in EDIT AUDIO mode, and recording is taking place, pressing PLAY punches out of record.

The PLAY button automatically lights when F.FWD or REW is pressed after CUE (item 4).

㉕ RECORD button
When pressed together with PLAY, enables record mode. When the deck is in EDIT AUDIO mode and is playing, it will drop into record upon hitting RECORD.

㉖ MEMO 1 and MEMO 2 keys
Used to set locations to which the deck will be autolocated when you press the LOC 1/2 key. Setting MEMO points at menus is explained in Section 10, page 10 • 1.

MEMO points can be fine tuned within 30 frames behind or ahead of the original point, as explained in Section 11, pages 11 • 1 and 2.

Turning off the deck does not clear MEMO points from memory.

㉗ LOC 1 and LOC 2 keys
Used to autolocate the deck to the MEMO 1 and 2 points, respectively.

㉘ MEMORY START key
This ensures instant transition from Stop to Play. When you press this key and locate the tape to the point where you want to let play start from, the first 3 seconds of digital audio are stored into a memory buffer ; so that, when you hit PLAY, audio is read from the memory instead of directly from the tape and there can be no delay between the time you hit PLAY and hear audio. See Section 11 for more information.

㉙ AUTO IN/OUT switch
Causes the deck to drop into record at the MEMO 1 point, and drop out of record at the MEMO 2 point. See pages 12 • 1 and 15 • 5 for details.

㉚ CHASE key
When this is pressed on, the DA-60 MKII constantly compares the timecode read off tape with the incoming timecode so that the deck operates in sync with external machines.

The DA-60 MKII offers two sync modes:

(1) Re-chase mode : In this mode, two timecodes, one from the master and one from a slave, are compared constantly, and each time the slave advances or delays with respect to the master, the slave is made to speed up or slow down to keep up with the master.

(2) Free run mode : In this mode, the slave chases and locks to the master only once ; thereafter the slave does not respond to the master and operates independently.

㉛ MONITOR select switch
This switch toggles between two monitoring options : TAPE and SOURCE, switching the headphones output, and also both the digital and analog outputs, to the tape or to the input.

㉜ VARI SPEED switch
Pressing this switch allows you to use the rotary dial to change the tape speed. See page 14 • 1 for details.

㉝ SET and RESET keys
SET is used to save your settings at menus, and RESET is used to cancel your settings.

㉞ AUTO CUE switch
Automatically cues the deck to "first frame of audio" (or the very fast musical note) in a program. See page 11 • 2 for details.

㉟ REHEARSAL key
Used when a trial punch in is required without actually recording on to tape (page 12 • 2). Also used for a trial Memory Start (page 11 • 1).

㊱ DATA switch and rotary dial
Used to select parameters from menus. Also, rotating the dial while holding SHIFT scrolls the menus.

☞ *While the VARY SPEED LED is on, the rotary dial is used to increase or decrease the play speed.*

㊲ Fs (sampling frequency) select switch
Selects a sampling frequency when recording from analog input.

㊳ REMOTE switch
Setting this switch to OFF selects the ACCESSORY 1 parallel data port for interface to external systems ; and setting to ON selects the RS-424 serial data port, instead.

> ***NOTE***
>
> *Setting the switch to ON disables all the controls on the DA-60 MKII.*

㊴ INPUTS level controls
Can operate only when the INPUT source select switch is at ANALOG position and the ANALOG INPUT switch (item 18) is at UNCAL position.

2-2. Rear Panel

㊵ DIGITAL IN
For connection to the XLR digital audio output of external equipment. Either the AES/EBU format data (IEC 958 TYPE I) or a consumer format data (IEC 958 TYPE II) can be plugged in ; the deck configures itself for the type of incoming data.

㊶ DIGITAL OUT
For connection to the XLR digital input of external equipment. Only the AES/EBU format data is available at this output.

㊷ 75 ohm ON/OFF switch
Setting this switch to ON position disables the WORD SYNC THRU jack by terminating the WORD SYNC IN jack in 75-ohm resistor. When using the THRU jack, set the switch to OFF position.

㊸ WORD SYNC jacks
Used for the deck to operate in sync with external machines by referencing to the Word Sync signal.

The IN signal is "echoed" out the THRU jack unless the 75 ohm switch is at ON. The THRU jack may be connected to a second slave machine.

㊹ AC IN
For connection of a supplied AC power cable.

㊺ ANALOG INPUTS
These XLR-type connectors accept balanced signals from external units.

Pin assignment : Pin 1 shield (GND), Pin 2 hot (+), and Pin 3 cold (-).

㊻ TIME CODE IN and OUT
For receiving and transmitting SMPTE/EBU timecodes.

㊼ VIDEO IN
Accepts a video composite or film sync signal from VTRs or film editing machines to which the DA-60 MKII will be slaved.

㊽ RS-422 port
This 9-pin D-sub connector is for serial interface to computers or controllers. The pin assignment is shown on the next page.

㊾ ANALOG OUTPUTS (XLR-type connectors)
For connection to the balanced analog input of external units.

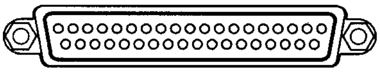
Pin assignment : Pin 1 shield (GND), Pin 2 hot (+), and Pin 3 cold (-).

㊿ ACCESSORY 1 port
This 37-pin D-sub connector is for parallel interface to the optional remote control RC-D6, or to computers or others. The pin assignment is shown on the next page.

(51) ANALOG OUTPUTS (RCA jacks)
For connection to the unbalanced input of external units, including monitor systems.

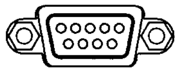
2-3. Pin Assignment

● **ACCESSORY 1 Port**



Pin	Signal
1	PLAY IN
2	F.FWD IN
3	REW IN
4	REHEARSAL IN
5	STOP IN
6	REC IN
7	CUE IN
8	FADER START IN
9	——
10	——
11	PLAY OUT
12	F.FWD OUT
13	REW OUT
14	STOP OUT
15	REC OUT
16	CUE OUT
17	START – ID OUT
18	——
19	STAND BY OUT
20	END OF TAPE OUT
21	——
22	START – ID WRITE IN
23	ID SEARCH NEXT IN
24	ID SEARCH PREVIOUS IN
25	——
26	——
27	——
28	——
29	——
30	——
31	——
32	——
33	——
34	——
35	——
36	GND
37	+5V

● **RS-422 Port**

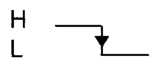


Pin	Signal
1	FRAME GROUND
2	TRANSMIT A
3	RECEIVE B
4	TRANSMIT COMMON
* 5	BRAKE
6	RECEIVE COMMON
7	TRANSMIT B
8	RECEIVE A
9	FRAME GROUND

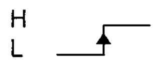
*) Pin 5 is used only when connecting to the TASCAM ES-61 edit controller.

• **Input** : To activate a function, the pin must be brought to ground potential for 50 msec or more.

Pin 8: FADER START



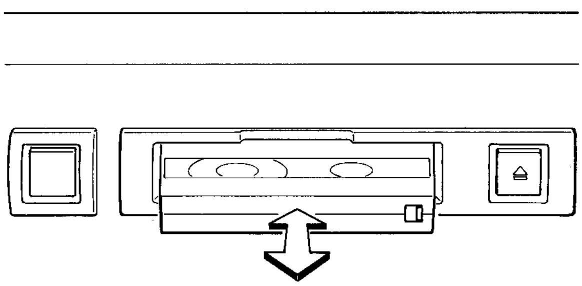
FADER STOP



• **Output** : Open collector. Maximum allowable current 100 mA
+5V supply : Maximum allowable current 0.3 A.

SECTION 3 : DAT CASSETTE TAPE

3-1. How to Load a DAT Cassette



☞ *Only when the DA-60 MKII is turned on, you can load DAT tapes.*

The hinged part of the cassette must go in first, with the clear window facing up. Similar to a VCR tape, the label surface of the cassette will be visible. When you encounter a slight bit resistance, push the cassette gently.

When a cassette is loaded, the CASSETTE IN LED will glow solid. This indicator will blink when you press EJECT, and will turn off when the cassette is ejected.

NOTES

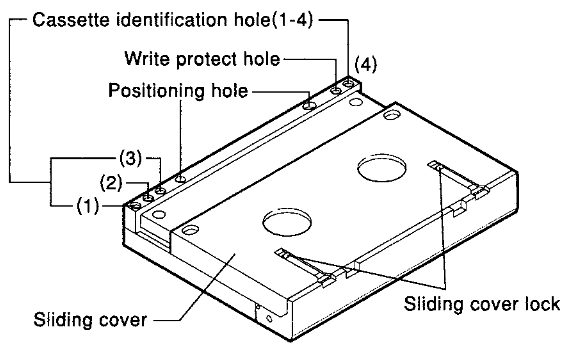
- *Cassette shells are designed so as to prevent touching the tape directly by hand.*
- *DAT cassettes can be loaded and unloaded only when the DA- 60 MKII is switched on.*
- *DAT cassettes record and play in one direction only. Do not load DAT cassettes upside down.*
- *DAT cassettes have a tape protection lid on the front edge to protect the tape. Do not open this lid forcibly, and do not pull the tape out from the cassette or touch it with your fingers.*
- *Be sure to replace DAT cassettes in their plastic cases for storage.*
- *Do not place DAT cassettes on a television, speaker or near equipment which could generate a magnetic field.*

Don't use 180-minute cassettes in the DA-60 MKII

The tape used in 180-minute cassettes is extremely thin and can cause winding problems, crimping, wrinkling, and other damage to data on tape.

3-2. Structure of DAT Cassettes

Bottom view



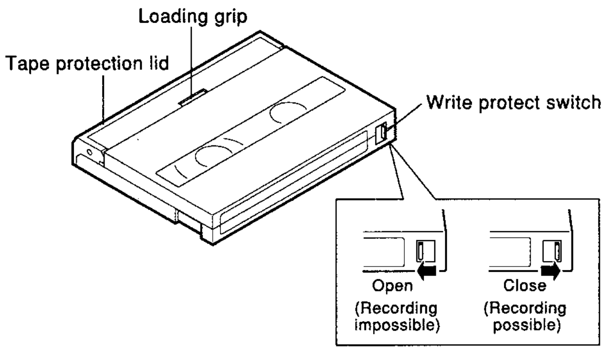
Identification Hole 1	2	3	Signified
X	X	X	Metal coating or equivalent/ 13 μm tape thickness
X	O	X	Metal coating or equivalent/ Thin tape
X	X	O	1.5 time track pitch/13 μm tape thickness
X	O	O	1.5 time track pitch/Thin tape
O	—	—	(Reserved for auxiliary tape type definitions)

Where : "O" = Open
"X" = Closed

Identification Hole 4

O = Pre-recorded tape sold by record companies
X = Blank tape

Top view



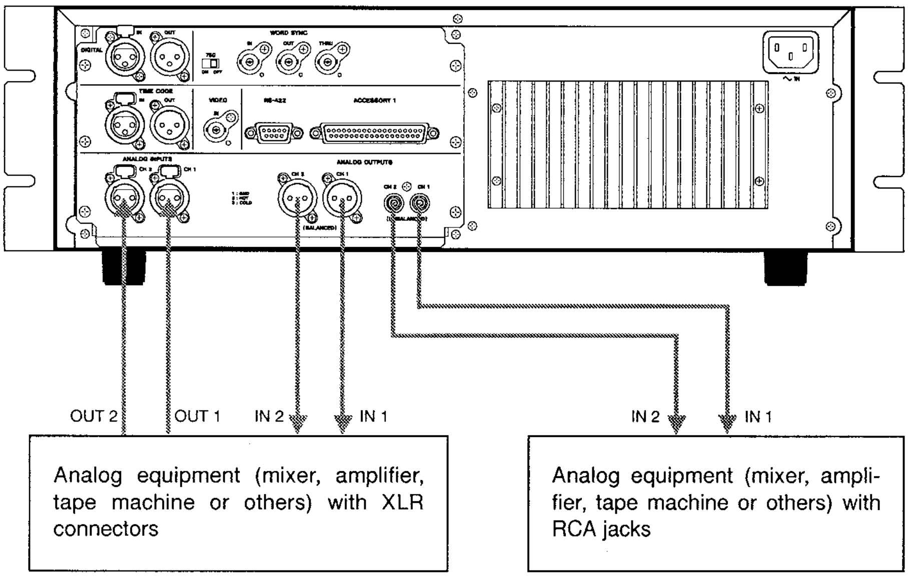
- Dimensions: 73 x 54 x 10.5 mm(W x D x H)
- Tape width: 3.81 mm

SECTION 4 : HOOKUP EXAMPLES

This section of the manual is only intended to provide basic hookup examples for you to get an idea of what you will need for your particular setup.

4-1. Connection to Analog Equipment

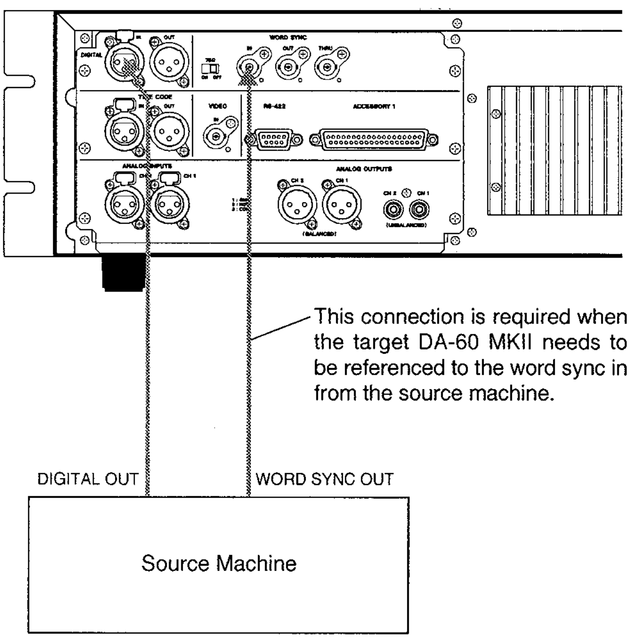
■ **Example : Recording and Playing Analog Audio Signals**



The INPUT switch should be set to ANALOG.

4-2. Connection to External Digital Equipment to Make a Digital Copy

■ **Example 1 : Using the DA-60 MKII as the Target Machine**

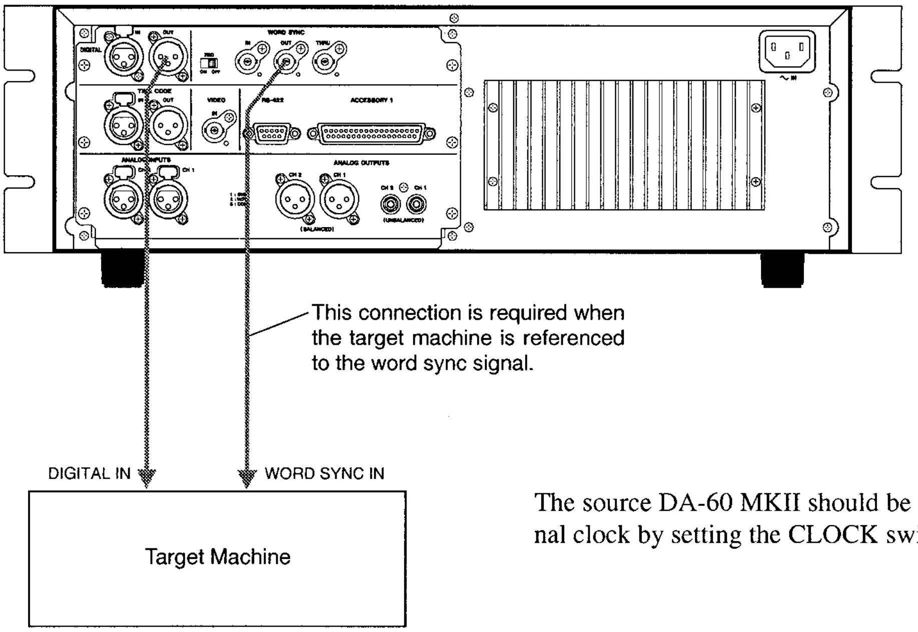


This configuration requires the following switch settings (on the front panel) :

- INPUT switch to DIGITAL :
- CLOCK switch to EXT ;
- EXT SYNC switch to DIGITAL, or to WORD, depending on the clock to which the source machine is referenced.

The DA-60 MKII is slaved to the clock derived from the digital in, or from the word sync in, and receives digital signal from the digital in.

■ **Example 2 : Using the DA-60 MKII as the Source Machine**



The source DA-60 MKII should be referenced to its internal clock by setting the CLOCK switch to INT.

NOTE

When making a digital copy, remember the following :

The digital output carries AES/EBU format data (IEC958 Type I), and whatever is written in the subcode area of the tape is NOT sent out.

If you want to copy Start IDs along with audio, connect the Start ID output available at pin 17 of the ACCESSORY 1 (D-sub, 37-pin) connector to Start ID input pin 22 of the identical connector on the target DA-60 MKII, or to an equivalent input if any other recorder is used as the target machine.

SECTION 5 : INITIAL SETTINGS

Before starting recording, you have to perform the following settings.

5-1. Selecting a Record/Write Mode

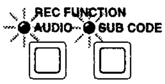
The DA-60 MKII offers three record/write modes : ASSEMBLE, EDIT AUDIO, and EDIT SUB.

ASSEMBLE mode allows both audio and subcode data to be recorded at one time. Use this mode when you are using a new blank tape.

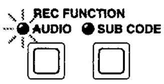
☞ *ABS time is automatically recorded. Recording of Start and End IDs is up to you (discussed later).*

EDIT AUDIO mode which allows you to edit audio data previously recorded in ASSEMBLE mode. The existing subcode data is not affected. This mode is typically used for punching in and out.

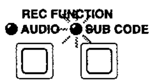
EDIT SUB mode allows only subcode data to be edited.



To select ASSEMBLE mode, press AUDIO and/or SUB CODE of REC FUNCTION so that both LEDs turn on.

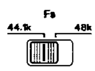


To select EDIT AUDIO mode, press AUDIO and/or SUB CODE so that the AUDIO LED turns on and the SUB CODE LED turns off.



To select EDIT SUB mode, press AUDIO and/or SUB CODE so that the SUB CODE LED turns on and the AUDIO LED turns off.

5-2. Selecting a Sampling Frequency



If you intend to record analog inputs on a new tape, set the Fs switch to the left 44.1 k position for CD production, or set to the right 48 k position for other applications.

This setting is required only when recording from the analog input. The DA-60 MKII configures itself for the correct sampling frequency when editing audio data previously recorded, recording from the digital source, or playing back a tape.

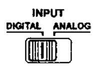
NOTE

To insure against any trouble caused by coexistence of different sampling rate information on a tape, if you want to use an old tape and overwrite whatever is previously recorded, erase the tape from start to end using a metal-tape bulk eraser.

If an appropriate eraser is not available, use this alternative way :

Immediately after inserting the tape into the DA-60 MKII, press and hold STOP until the button lights, so the deck cannot read sampling rate information from the tape, and recording can be made at a sampling rate you select with the Fs switch.

5-3. Selecting Analog or Digital Input



Depending on the source connection, set the INPUT switch to DIGITAL or to ANALOG.

If you select the digital input, the indicator "DIGITAL" will light up on the display when a digial signal is actually fed in.

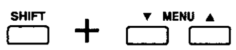
The DA-60 MKII can accept both the AES/EBU and consumer format data, configuring itself for an incoming format of data.

5-4. Selecting a Copy ID

Copy ID is the flag which determines how many generations of copy you can add to the original digital recording. (This ID is written to ID6 in the main dada area of the tape.)

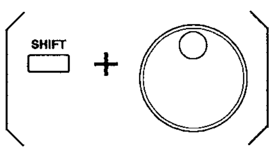
☞ *This setting is required and valid only when recording consumer format digital data following the SCMS (Serial Copy Management System).*

To Select a Copy ID

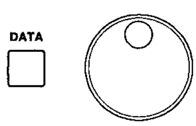


1 Hold SHIFT and press MENUs until the menu "4 coPy" shows on the left hand side of the display.

4 coPy id-6 00

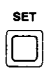


Suggestion : An alternative is to rotate the DATA dial while holding SHIFT.



2 Press DATA, and rotate the dial. The blinking number will change as follows :

- id-6 00 : Indefinite number of generations of copy can be added.
- id-6 10 : No digital copy can be made.
- id-6 11 : Only one generation can be added. To make a copy of a copy is impossible.



3 Press SET to save your selection.

To switch the display back to show timecode numbers, press either MENU key.

- The above setting has nothing to do with recording/copying AES/EBU format

5-5. Selecting a Sync Signal

To let the DA-60 MKII play or record in sync with other digital audio machines, it must be referenced to an incoming clock.

Depending on the clock to which the DA-60 MKII is to be referenced, set the CLOCK and the EXT SYNC switches as follows :

CLOCK	EXT SYNC	S2 (see page 7•2)	Indication	Referenced to
INT	—	—	INT LED*	Internal clock
EXT**	DIGITAL	—	DIGITAL	Clock derived from the digital in (INPUT switch at DIGITAL)
	WORD	—	WORD	Clock derived from the word sync in
VIDEO	—	VIDEO	VIDEO	Clock derived from the video composite signal coming into the video in
	—	FILM	VIDEO	Film sync signal coming into the video in

* The INT LED will light when :

1) the CLOCK switch is set to INT and the clock is correctly being generated OR

2) although the CLOCK switch is set to EXT or to VIDEO, the clock is not coming in, and the internal clock is activated instead.

** When the DA-60 MKII is in play mode and is referenced to an external clock (either derived from the digital in or from the word sync in), a maximum of +/-12.5% speed variation on external transports does not release the sync lock.

When recording onto the DA-60 MKII, you can select either the limits of +/-100 ppm or of +/-12.5% at a menu ("14 bAnd"), as discussed later, page 17 • 4.

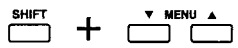
5-6. Pre-emphasis On/Off

This setting is required only when recording analog audio inputs.

The DA-60 MKII converts analog signals into digital format, and records encoded signals (a series of numbers) on tape. Before converted into digital format, analog signals are pre-emphasized to boost high frequencies ; and during play these are cut (de-emphasized) to minimize high frequency noise, thus improving signal-to-noise ratio.

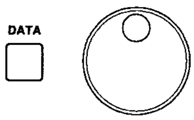
The DA-60 MKII has the capability of turning the emphasis circuit on or off to accommodate various situations. Your setting is recorded on tape along with audio data; and, playback is de-emphasized or not depending the setting recorded on tape.

To Switch the Pre-emphasis On or Off :



1 Hold SHIFT and press MENUs until the menu "3 PrEEP" (pre-emphasis) shows on the left hand side of the display.

3 PrEEP oFF



2 Press DATA, and rotate the dial to change the blinking "oFF" to "on" or vice versa, as required.

3 Press SET to save your selection.

If you have selected "on", the display will show "EMPHASIS".

☞ *There are more other menu-controlled functions. Set also them to meet your requirements by referencing to Section 17.*

SECTION 6 : RECORDING AUDIO INPUT

If you want to perform a punch-in recording, go to the section on Punch In and Out, page 12 • 1.

Before initiating audio recording, consider the Subcode Data recording possibilities and limitations (discussed later).

1 Check that the following are correctly set as per instructions given in the previous section of this manual :

(1) Record mode : Select the ASSEMBLE mode for recording on a new, blank tape.

(2) Sampling frequency

(3) Input source

(4) Copy ID

(5) Sync reference signal

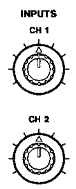
(6) Pre-emphasis

2 Set the recording level as follows :

(1) Press MONITOR until "SOURCE" shows in the display. The audio outputs (Phones, Analog, and Digital) are now all switched to carry the input signal.

(2) Set the ANALOG INPUT switch to UNCAL, and play the source and adjust the INPUTS level controls until the meters peak at "0".

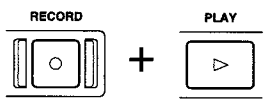
In the CAL position a +4 dBm signal causes the meter to read -16 dB. The level controls have no effect on the reading.



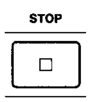
3 Press MONITOR once more to have "TAPE" appear on the display so you can monitor recording off tape. The peak meter also then registers the level off tape, not the input level.

Or, leave MONITOR switched to SOURCE if you want to monitor the input signal.

In addition if the deck is in EDIT AUDIO mode, the monitor automatically switches to "SOURCE" when recording starts. And during recording you cannot switch it to the tape.



4 When everything is ready, press RECORD and PLAY together to start recording.



5 Press STOP to terminate recording.

The tape will automatically move backward over the length of about 1.5 seconds before stopping (this occurs only in ASSEMBLE mode) ; so that, when you start a new recording, there is no break in the sub-code data.

SECTION 7 : STRIPING A TAPE WITH TIMECODE

To have the DA-60 MKII sync up to other audio recorders or VTRs, the same type of timecode must be registered on their tapes. You can stripe timecode on a DAT tape either by using the internal timecode generator or by copying a timecode existing on an external tape. This unit converts timecode (whether it comes from the internal generator or an external tape) to professional DAT format and writes it in a sub-code (non-audio) area of the tape.

Precautions and Recommendations

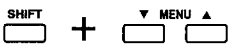
- The same type of timecode must be recorded on both the master and slave tapes or else erratic synchronization and autolocation will occur.

- When recording program material, allow a sufficient leader tape ahead of each of them. Similarly, allow a sufficient length of no-audio section after the end of the last program on every tape.

7-1. Getting Ready to Record Timecode

To access timecode-related menus you fisrt have to open a "tc" menu.

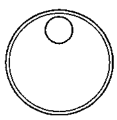
Opening the menu "tc"



1 Hold down SHIFT and press MENUs until the menu "tc" appears on the left hand side of the display, which looks like this :

-tc- cLoSE

2 Press DATA. The "cLoSE" will start blinking.



3 Turn the rotary dial to change the "cLoSE" to "oPEn."

4 Press SET to save the setting.

From now on you can access any of the timecode-related menus at any time you need. They include the following menus that pertain to group 1 :

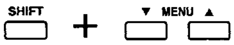
- EXT TC
- GEN TC
- GEN TC SET
- CHASE OFFSET
- U-BIT EXT
- U-BIT

See also Section 17.

Selecting a Type of Timecode

It is imperative that one and the same type of timecode is used throughout your system.

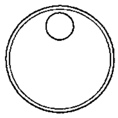
The DA-60 MKII is factory preset to 29.97 dF. To select other types of timecode :



1 Hold down SHIFT and press MENUs until a "rEFtc" (reference timecode) menu appears on the left side of the display, which looks like this :

22 rEFtc   2997 dF

2 Press DATA. The "2997 dF" will start blinking.



3 Turn the rotary dial to select the desired type of timecode among the following (in addition to "2997 dF") :

- "30 ndF" (for SMPTE 30 non-drop frame code)
- "2997 ndF" (for SMPTE 29.97 non-drop frame code)
- "25 Ebu" (for EBU 25 frame code)
- "24 Fil" (for FILM 24 frame code)
- "30 dF" (for SMPTE 30 drop frame code)

4 Press SET to save your setting.

The associated indicator (30, 29.97, etc) will blink in the display if a different type of code from the one you just selected comes in from external units.

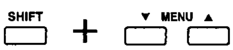
If you intend to slave the DA-60 MKII to a film sync signal (square wave, TTL level) plugged into the video input, dip- switches S-2 (located on the SYNC PC Board) must be set as follows :

	1	2	3	4			1	2	3	4
ON			–	–	⟶	ON		–		
OFF	–	–				OFF	–		–	–
VIDEO SIGNAL						FILM SIGNAL				

> ***WARNING***
>
> *This setting must be done only by a qualified service person.*

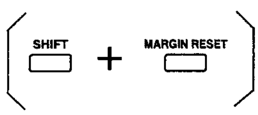
Selecting a Generator Mode

The internal timecode generator is factory preset to "rEc run" mode. To check or change the current generator mode :



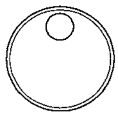
1 Hold down SHIFT and press MENUs until a "tcGEN" menu appears on the left hand side of the display, which looks like this :

24 tcGEN   rEc run



Suggestion : Alternatively, you can hold SHIFT and press MARGIN RESET to access the "tcGEN" menu.

2 Press DATA. The "rEc run" will start blinking.



3 Turn the rotary dial and the following modes will show in sequence :

- rEc run : In this mode the internal timecode generator starts generating timecode from a user selected time point and recording it on tape when the transport enters either ASSEMBLE or EDIT SUB record mode.

- FrEE run : This mode is independent of transport mode, and upon selecting this mode the internal timecode generator immediately starts generating timecode from a user selected time point, as confirmed by the message "FREE RUN" being flashing.

  To stop the generator, select any other mode than "FrEE run".

  In "FrEE run" mode, timecode on tape cannot be played back.

  You can view timecode numbers on the right hand side of the display as they are generated and recorded if you press MENU to have a "■ GEN TC" indicator appear.

- J-Sync (jam sync) : This mode is used to add timecode to the end of the prerecorded code on tape in such a way that there will be continuous, uninterrupted timecode on tape.

- oFF : Select this mode when recording timecode coming from the exterior on a DAT tape.

4 Press SET to save your setting.

7-2. Striping Timecode using the Internal Generator

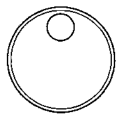
From a user selected time point onward

1 Make sure that "rEc run" is selected at the "24 tcGEn" menu, as discussed above.

2 Press MENU until "■ GEN TC SET" lights up in the display. Below the indicator will show the current start time.

3 Press DATA. The hour digits will start blinking.

Each time you press DATA, the next lower (right) two digits will blink.

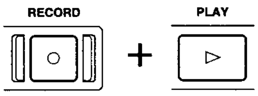


4 When the digits you want to set are blinking, turn the rotary dial so that the desired number appears.

You can press RESET to clear the blinking digits to 00.

5 When you are satisfied with all the hour, minute, second and frame displays, press SET.

6 Enter the ASSEMBLE mode if you intend to record audio at the same time as timecode. Or, if you want to record timecode only, enter the EDIT SUB mode. Note, however, in EDIT SUB mode, timecode cannot be recorded on a non-recorded blank (neither audio nor sub-code data) section of the tape.



7 Hold RECORD and press PLAY to initiate timecode recording.

You can see timecode numbers as they are recorded on tape.

Suggestion : When recording timecode with the generator switched to its "rEc run" mode, the code is sent out of the TIME CODE OUT jack, allowing it to be recorded on external machines as well. But don't use this local striping capability with VTRs since the internal generator is not timed to coincide with rising edges of the video frame.

To Add Timecode to the End of the Prestriped Code

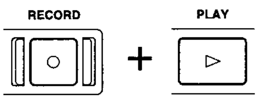
To record additional timecode on a tape which is previously striped with code up to an intermediate point, proceed as follows :

1 Select "J-Sync" mode at the "24 tcGEn" menu.

2 Locate the deck to a point 2 seconds or more lower than the end of the existing code.

This is absolutely necessary for the deck to read the existing code and generate code starting from the correct number.

3 Enter the ASSEMBLE mode to record audio at the same time as timecode. Or, select the EDIT SUB mode to record timecode only. Speaking of remembering, in EDIT SUB mode, timecode cannot be recorded on a non-recorded blank (neither audio nor sub-code data) section of the tape.



4 Hold RECORD and press PLAY to let timecode recording start.

- If no timecode is prerecorded on the tape, the generator starts from a user selected time point overriding the J- Sync mode.

7-3. Copying Timecode from External Units

To record external timecode onto the DA-60 MKII, the internal generator must be in "oFF' position, as selected at the "24 tcGEn" menu.

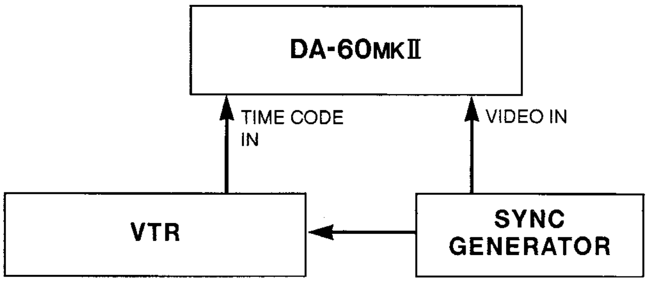
☞ *You cannot copy timecode from ATRs onto the DA-60 MKII. If you want, you can copy only audio data from an ATR by letting it run in sync with the DA-60 MKII after having recorded timecode on the tape in the DA-60 MKII.*

To Copy Timecode from VTRs

1 Check to see that both the DA-60 MKII and the VTR are turned off.

2 Make the following connections :

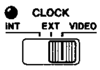
- Connect the timecode from the VTR to the TIME CODE IN jack on the rear of the DA-60 MKII.
- Connect composite video signal from a sync or pattern generator in use to the VIDEO IN jack on the rear of the DA-60 MKII.



3 If you intend to copy both audio data and timecode at the same time (otherwise, skip to step 4) :

- Connect the audio output of the VTR to the DIGITAL IN or to the ANALOG INPUTS jacks on the rear of the DA-60 MKII depending on the audio output.
- Set the INPUT select switch to ANALOG or to DIGITAL depending on the connection.
- Set the CLOCK switch to EXT when copying from the digital input.

4 Turn on the DA-60 MKII, and also the VTR.



5 Set the CLOCK switch to VIDEO.

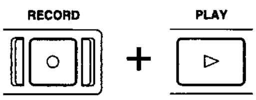
- To copy digital audio data at the same time as timecode, the switch must be set to EXT, as said in step 3.

6 Enter the EDIT SUB record mode.

- To copy audio data (digital or analog) at the same time as timecode, select the ASSEMBLE mode.

7 If your VTR has a variable speed function, check to see that it is disabled.

8 Put the VTR into play mode.



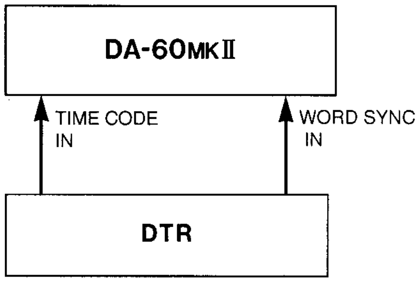
9 Put the DA-60 MKII into record mode by holding RECORD and pressing PLAY.

Copying Timecode from DTRs (Digital Tape Recorders)

1 Check to see that both the DA-60 MKII and the DTR are turned off.

2 Make the following connections :

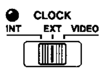
- Connect the timecode from the DTR to the TIME CODE IN terminal on the rear of the DA-60 MKII.
- Connect the word sync out from the DTR to the WORD SYNC IN jack on the DA-60 MKII.



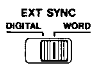
3 If you intend to copy audio data at the same time as timecode (otherwise, skip to step 4) :

- Connect the audio output from the DTR to the DIGITAL IN jack on the DA-60 MKII.
- Set the INPUT select switch to DIGITAL.

4 Turn on the DA-60 MKII, and also the DTR.



5 Set the CLOCK switch to EXT.



6 Set the EXT SYNC switch to WORD.

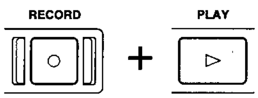
- Only when your DTR has no word sync output, set the switch to DIGITAL so the DA-60 MKII can be referenced to the clock derived from the digital in.

7 Enter the EDIT SUB mode.

- Or, select the ASSEMBLE mode to copy both audio and timecode at the same time.

8 If your DTR has a variable speed function, check to see that it is disabled.

9 Put the DTR into play mode.



10 Put the DA-60 MKII into record mode by holding RECORD and pressing PLAY.

7-4. Recording ABS (Absolute) Time

ABS time indicates the total time elapsed from the beginning of the tape. It is registered automatically (in ASSEMBLE or in EDIT SUB mode), but be aware of the following points :

- When using an unrecorded, blank tape, be sure to rewind it all the way to the beginning before starting recording. Otherwise ABS time is not recorded.
- ABS time is sequentially recorded when the deck starts recording (in ASSEMBLE or in EDIT SUB mode) from a point where ABS time is already recorded. Even the EDIT SUB mode does not allow recording ABS time on a non-recorded blank section of the tape.
- During recording ABS time you can see it increment on the left hand side of the display if you select "AbS" at a "21 Pbtc" menu.
- No ABS time is displayed during play.

SECTION 8 : PLAYBACK

8-1. Selecting a Playback Timecode Reference

During play the DA-60 MKII converts a professional DAT timecode or the ABS time data into SMPTE/EBU timecode. This conversion is controlled from a "21 Pbtc" menu. This menu offers three optional modes :

- "Auto" : In this mode the professional DAT timecode registered on tape or, if it is not available, the ABS time data is converted into SMPTE/EBU timecode.

- "tc" : This mode converts the professional DAT timecode into SMPTE/EBU timecode.

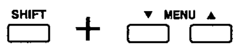
- "AbS" : This mode converts the ABS time data into SMPTE/EBU timecode.

The DA-60 MKII achieves such operations as autolocation and synchronization by referring to the thus converted SMPTE/EBU timecode, the type of which depends on your selection at the menu "22 rEFtc" (page 7 • 2).

- If the professional DAT format data on the tape originated from the 29.97 f/s time code (drop or non drop) and another type of SMPTE/EBU timecode is selected before play, then the display for timecode numbers could "jump" at the boundary of "0" hour or could read "24" hours or more. This is because of discrepancy between the timecode numbers and the actual time, and is not due to any trouble of the DA-60 MKII.

- Among consumer DAT recorders there are some which cannot record ABS time data correctly. If a tape recorded on such machines is loaded on the DA-60 MKII and the frame digits blink on the left side of the display, it shows that the ABS time data on the tape is not accurate enough for the correct synchronization or autolocation. If this is the case, record timecode afresh in EDIT SUB mode.

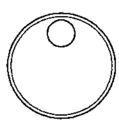
To select a playback timecode reference



1 Hold down SHIFT and press MENU until a "21 Pbtc" menu shows on the left hand side of the display.

21 Pbtc Auto

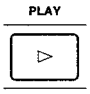
2 Press DATA. The "Auto" will start blinking.



3 Turn the rotary dial to change the "Auto" to "tc" or to "AbS", as required.

4 Press SET to save your selection.

8-2. Playback

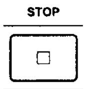


1 Press PLAY and playback starts.

- Upon pressing PLAY the monitor automatically switches to "TAPE".
- If STANDBY is pressed on, the head drum is in motion, so you'll hear audio immediately after hitting PLAY.

  If leaving the deck in STANDBY mode for about 3 minutes, the mode is automatically cancelled to avoid tape and head wear.
- You can hear the tape play at high speeds if in CUE mode.
- During play the following will show on the display :

  ○ The current program number
  ○ Timecode numbers
  ○ EMPHASIS indicator (if the recording was emphasized)



2 To stop playback, press STOP.

SECTION 9 : RECORDING SUB-CODE DATA

9-1. About DA-60 MKII Sub-codes

DAT recorders are capable of recording sub-codes apart from the audio data. The DA-60 MKII can handle the following sub-code data :

- Start ID
- End ID
- Program Number
- ABS Time
- Professional DAT Timecode

9-2. Recording/Erasing Start IDs

Recording Start IDs

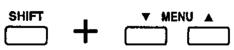
Start IDs are used to mark the beginning of a program or of a section, so that you can quickly move to any start point for play. They can be recorded automatically or manually.

> ***NOTE***
>
> *Any transport controls except for STOP cannot operate when a Start ID is being registered on tape.*

Automatic Recording (along with Audio)

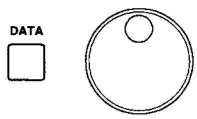
Perform the following steps before starting audio recording.

1 Select the ASSEMBLE mode.



2 Hold SHIFT and press MENU to access a "5 At-id" menu.

5 At-id -54db



3 Press DATA and rotate the dial to select a level at each occurrence of which a Start ID will automatically be registered.

Options available :
- "oFF" : No Start ID is automatically registered.
- "-54 db"
- "-60 db"
- "-66 db"

A Start ID is registered each time a sound louder than the selected level is fed into the deck following a period of silence or of a sound lower than the selected level at least 2 seconds long.

The AUTO ID indicator will light up in the display unless "oFF" is selected.

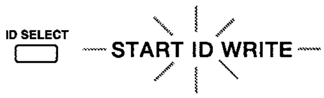
4 Press SET to save your setting.

- You can switch the Auto ID function on or off by holding SHIFT and pressing ID SELECT, without having to access the menu.

Manual Recording (along with Audio or during Play)

> ***NOTE***
>
> *You cannot record a Start ID overlapping the existing ID. Previously erase unnecessary marks, as explained later.*

1 Select the ASSEMBLE mode or the EDIT SUB mode.



2 Press ID SELECT until the "START ID WRITE" indicator lights up in the display.

3 Start recording if in ASSEMBLE mode, or start playing if in EDIT SUB mode, and hit EXE at the desired moment. The "START ID WRITE" indicator will blink, showing that a Start ID is being recorded. This takes about 9 seconds.

- To fine tune the point where you want a Start ID to start being marked from :

  1. During record or play, hit MEMO (1 or 2) at a tentative point.
  2. Stop the tape, then follow the same procedure as for "MEMORY START Play" (11-1 through 11-3 on pages 11 • 1 and 2).
  3. Enter the EDIT SUB mode.
  4. Press ID SELECT until the "START ID WRITE" indicator light up in the display.
  5. Press EXE and a Start ID is written from the fine tuned point on.

To Erase Start IDs (during Play or Stop)

1 Select the EDIT SUB mode.

2 Press ID SELECT until the "START ID ERASE" indicator lights up in the display.

3 When the tape is stopped at or playing a higher point than the Start ID you want to erase, press EXE. The tape rewinds to the beginning of the ID, and this is erased while the tape is playing.

While a Start ID is being erased, the "START ID ERASE" indicator will flash.

> ***NOTE***
>
> *When erasing a Start ID, the program number is also erased. Renumbering program numbers is explained below.*

9-3. Program Numbers

Program numbers indicate the position of each program in a sequence. They are automatically registered at the same as Start IDs.

Numbering Programs

When recording Start IDs (automatically or manually), program numbers are also recorded in sequence to existing numbers. When a Start ID is automatically registered by letting audio record start from the beginning of the tape in ASSEMBLE mode, "001" is assigned to the first program.

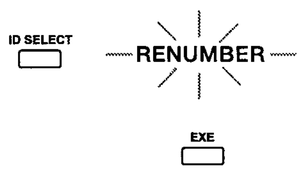
- If you want to start audio recording mid-way through a tape, first play the previous program so the deck can read and display its number. Otherwise, no program number is registered.
- Recording the desired program number is explained later, page 10 • 2.
- Program numbers can recorded from 001 up to 799. But the PGM NO. display shows only the right two digits (the hundreds are not shown).

Erasing Program Numbers

When you erase a Start ID, the program number also is automatically erased.

9-4. Renumbering Program Numbers

If you erase or add Start IDs, the program numbers on the tape will become out of order. To put them back into order, perform renumbering as follows :



1 Enter the EDIT SUB mode.

2 Press ID SELECT until the "RENUMBER" indicator lights on the display.

3 During play or stop, press EXE.

The tape will rewind, and program number "1" will be recorded at the first Start ID the tape finds. When the first program is numbered (this takes about 9 seconds), the tape will be automatically located to the next Start ID, and this will be numbered "2". This process continues until all the existing Start IDs have program numbers in proper order. When all renumbering is complete the tape will automatically rewind, stopping at the beginning of the tape.

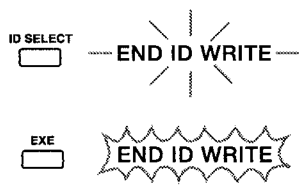
9-5. Recording/Erasing an End ID

An End ID identifies the end of the audio recorded section of a tape When fast forwarded, the tape will automatically stop at the beginning of the End ID mark, the "END ID" indicator appearing on the display.

If the tape encounters an End ID mark during play, it will rewind, stopping at the beginning of the tape.

> ***NOTE***
>
> *You can write the End ID mark only during audio recording in ASSEMBLE mode.*

Recording an End ID



1 Select the ASSEMBLE mode.

2 Press ID SELECT until the display shows "END ID WRITE".

3 Start audio recording, and terminate recording by pressing the EXE key.

The "END ID WRITE" indicator will blink on the display showing an End ID mark is being recorded. After 9 seconds the indicator will go out and the tape will automatically rewind, stopping at the beginning of the End ID mark just written.

Erasing the current End ID

The current End ID mark is automatically erased when a new audio reading is added to the end of the existing audio recordings.

SECTION 10 : AUTOLOCATION FUNCTIONS

You can quickly move to specific points on the tape, points stored in memory. Also, you can tell the DA-60 MKII to find the beginning of specific programs or the ID SEARCH function allows you to skip forward or back to other programs.

10-1. Setting Locations

"On the fly"

You can specify two points on the tape during record or play.

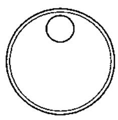
Simply hit MEMO (1 or 2) at the desired moment. The time point shown at that moment on the right hand side of the display is read into the corresponding memory. Each time either MEMO is pressed, a new memory point is established, erasing the previous memory in that register.

- Hitting MEMO during stop is also effective.

Setting locations at menus

1 Press MENUs until "MEMO 1" (or "MEMO 2") shows in the display. The current memory point will show on the right hand side of the display.

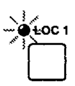
2 Press DATA until the number you want to change starts blinking (H, M, S or F).



3 Enter the desired number by turning the rotary dial.

4 When all digits are entered and they represent the desired locating point, press SET.

10-2. Autolocating to MEMO points

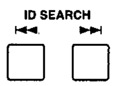


During stop or play, press LOC 1 to let the tape autolocate to, and stop at, the MEMO 1 point. Or, press LOC 2 to let the tape autolocate to the MEMO 2 point.

- Auto Play Function : Pressing PLAY after LOC will cause the tape to automatically start playing as soon as the location point is reached. MEMORY START must be switched Off.

All MEMO points (whether set at menus or captured on the fly) are automatically saved for later use when switching the power off.

10-3. Locating to a Start ID



During stop or play, press the forward ID SEARCH key to locate the tape to the next Start ID ; or press the backward ID SEARCH key to locate the tape to the previous Start ID.

Each time you press them, the tape will be located to the next or the last Start ID point. To skip several programs, press the key repeatedly. Each time you press the key, the number in the PGM NO. display window will increment. The "-" or "+" indication before the number shows the direction in which you are locating the tape.

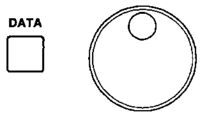
- The DA-60 MKII goes into STANDBY at the end of each search operation.
- During the search process, the REW or the F FWD button will light.
- Auto Play Function : Pressing PLAY after search operation starts will cause the tape to automatically start playing at the end of search operations. MEMORY START must be switched Off.

10-4. Program Number Search

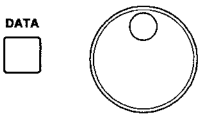
1 Press MENU until the display looks like this :

```
P-001      123 ←— Current program number
  A B       C
```

(A, B and C shown above are for an explanation.)

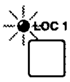


2 Press DATA. The lower two digits (marked B) will start flashing, so change them to the desired numbers (00 to 99) by rotating the dial.



3 Press DATA once more and the highest digit (marked A) will start flashing, so change it to the desired number (0 to 7) by rotating the dial.

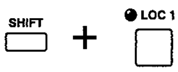
4 Press SET to save your setting.



5 Press LOC 1 and the deck will be autolocated to the beginning of the program just specified.

- Auto Play Function : Pressing PLAY after LOC 1 causes the tape to automatically start playing after completing autolocation.
- If MEMORY START is pressed on, 3 seconds of audio will be stored into a memory buffer after completing autolocation. For details see Section 11.

■ An alternative is to :

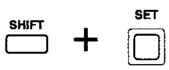


① Hold SHIFT and press LOC 1. The "P(rogram)" menu will appear.

② Enter the desired program number with the rotary dial, as explained above.

③ Press LOC 1 (directly, without pressing SET) and the deck will start autolocation.

■ Another Use of the "P" Menu — Assigning the desired program number when registering a Start ID



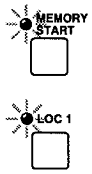
After completing step 4 above, hold SHIFT and press SET. The program number entered will show at the right (marked C) and also in the PGM NO. display window. Then, the deck recognizes this number as the current program number and, upon registering a Start ID, the next number is registered (e.g. 100 if 99 is entered).

SECTION 11. MEMORY START OPERATIONS

MEMORY START stores digital audio into a memory buffer over the length of about 3 seconds from a cue point on. This ensures an instantaneous, tight start because audio is read from the buffer not directly from the tape when you hit PLAY.

If you want to store audio from an AUTO CUE point on, first select a trigger level of the auto cue circuit, as explained under the corresponding heading, page 11 • 2.

11-1. To Store Audio into a Memory Buffer



1 Press MEMORY START. Its LED will turn on.

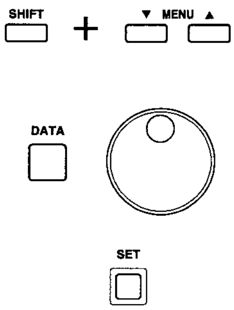
2 Have the deck autolocate to the desired memory point by pressing LOC 1 or 2. Or, use the ID SEARCH function to access the beginning of the desired program.

3 After completing autolocation, the deck enters play mode, storing audio data into a memory buffer, as confirmed by the MEMORY START LED flashing. About 3 seconds later, the LED glows solid and the tape rewinds, parking at the cue point in Memory Start Standby mode, as confirmed by the STOP button being lit and the PLAY button flashing.

11-2. To Audition the "Buffered" Audio

Pressing REHEARSAL while in Memory Start Standby mode allows you to hear audio for a user selected time (see below) as it is read from memory, not directly from the tape.

To Set the "Rehearsal" Time of Memory Start :



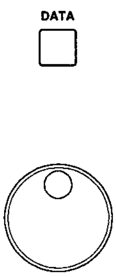
1 Hold SHIFT and press MENUs to access a rehearsal time menu which looks like this :

7 rEH-t   2000 ms

2 Press DATA and rotate the dial to display the desired time (from 100 to 2500 ms, in 100 ms steps).

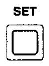
3 Press SET to save the setting.

11-3. To Trim Your Memory Start Point



1 Press DATA while in Memory Start Standby mode. The tape will play a loop of about 200 msec over and over. This loop precedes the beginning of the buffered 3 seconds of audio. You'll notice that "00" is flashing in the PGM NO. display window.

2 Rotate the dial to the left or right. The playing loop will move back or forward in 1 frame steps, up to 30 frames ahead of or behind the original point, as confirmed by the PGM NO. display.



3 Once the memory start point is trimmed to your satisfaction, press SET. Repeat play stops, the new memory start point is saved and the previous memory is erased.

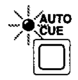
- If you press RESET instead of SET, repeat play stops and the original memory start point is retained.
- If you press a MEMO after SET, the memory start point just trimmed is stored into the corresponding register.
- The frame numbers you see in the PGM NO. display window are those of ABS time. But upon pressing SET, the ABS time data is converted to a timecode selected at the menu "22 rEFtc" (page 7 • 2).

> ***NOTE***
>
> *You cannot trim a memory start point when a "7 rEH-t" menu is shown.*

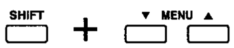
11-4. Auto Cue Trigger Level

If AUTO CUE is on, the deck starts storing 3 seconds of audio upon encountering a sound louder than a user selected level after passing by a point 1 second lower than a Start ID.

You can set the sensitive level of the Auto Cue circuit to -54, -60 or -66 dB, depending on the type of material you're working on.

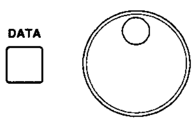


1 Press AUTO CUE and its LED will light.

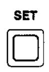


2 Hold SHIFT and press MENU to access an Auto Cue menu which looks like this :

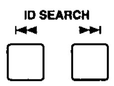
6 AtcuE -54 db



3 Press DATA to let the number display at the right start blinking, then turn the dial to enter the desired level number.



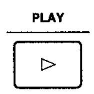
4 Press SET to save your selection.



5 Press MEMORY START to have its LED light, and let the deck autolocate to a Start ID using the ID SEARCH function. When a sound louder than the selected level is encountered after passing by a point 1 second lower than the Start ID, 3 seconds of audio are stored into a memory buffer and the deck parks in Memory Start Standby mode.

- If no sound louder than the selected level is encountered within 5 seconds after passing by a point 1 second lower than a Start ID, the deck will rewind the tape to the beginning of the Start ID, storing 3 seconds of audio from that point on.

11-5. Executing a Memory Start Play

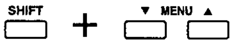


Press PLAY while in Memory Start Standby mode.

SECTION 12 : PUNCH IN AND OUT

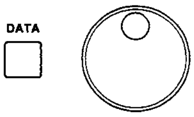
The DA-60 MKII's editing possibilities include click-free dropping into and out of record with an inbuilt crossfade action. The crossfade time is selectable at a menu.

12-1. To Select a Crossfade Time



1 Hold SHIFT and press MENU to access a Fade menu which looks like this :

9 FAdE 10



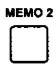
2 Press DATA and rotate the dial to change the flashing current corssfade time (unit : ms) to 10, 50 or 100.

3 Press SET to save the setting.

12-2. Automatic Insertion

Setting Punch In and Out Points

1 Load the point where you want the deck to drop into record into the MEMO 1 register, as explained in Section 10.



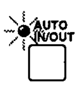
2 In a similar way, load the point where you want the deck to drop out of record into the MEMO 2 register.

- You can trim the punch in and out points using the same procedure as for the memory start point (pages 11 • 1 and 2).

NOTE

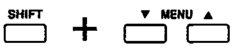
There must be at least 5 frames between the punch in and out points.

3 Enter the EDIT AUDIO mode.



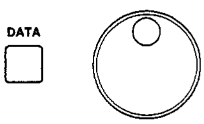
4 Press AUTO IN/OUT. Its LED will light.

Setting a Preroll Time



5 Hold SHIFT and press MENU to access a preroll time menu which looks like this :

13 PrE-r 5

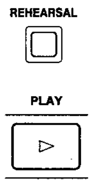


6 Press DATA, and the number display at the right starts flashing, and rotate the dial to enter the desired time (0 to 15 seconds).

7 Press SET to save the setting.

- When controlling the DA-60 MKII from P-2, the deck configures itself for an incoming preroll time.

Rehearsal



8 Press REHEARSAL, and the tape will fast wind, stopping the preroll time short of the punch-in point, and the AUTO IN/OUT LED will start flashing.

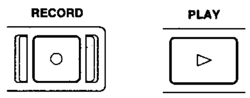
9 Press PLAY to have the tape start playing, the AUTO IN/OUT LED glowing solid.

- The monitor will automatically switch to SOURCE when the punch-in point is reached.
- The monitor will switch back to TAPE when the punch-out point is reached.

After about 3 seconds of postroll, the tape will rewind, and stop at the preroll start point, and the AUTO IN/OUT LED will start flashing again.

You can rehearse your punch-in as many times as you need without destroying the original take at all.

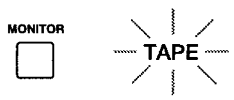
Committing the insertion to tape



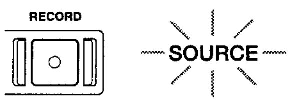
10 Press RECORD and then press PLAY. The same sequence as you have anticipated during Rehearsal will take place.

12-3. Manual Punch In and Out

1 Enter the EDIT AUDIO mode.

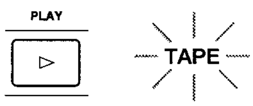


2 Press MONITOR until "TAPE" shows on the display.

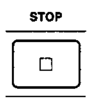


3 Start playback from a point lower than the expected punch-in point and, the instant this point is reached, hit RECORD. The monitor will switch from TAPE to SOURCE at the same time as initiating record.

- The MONITOR switch cannot operate while in record mode.



4 Hit PLAY when the point where you want to punch out of record is reached. The monitor will switch back to TAPE, allowing you to check if the new recording is smoothly followed by the previously recorded section.



5 Press STOP to stop the postroll.

SECTION 13 : FADE IN AND OUT

You can have the DA-60 MKII fade in and out of play or record ("A"-weighted).

> ***NOTE***
>
> *It is necessary to have the MEMORY START LED light up for playback to fade in and out.*

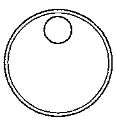
13-1. Setting the Fade In/Out Time

1 Using the MENU keys have the "FADE IN" and the "FADE OUT" indicators light up on the display, and the current setting will show on the right hand side of the display, like this :

in 3 out 5

2 Press DATA, and the fade in time display will flash, and rotate the dial to enter the desired time (0 to 9 seconds).

Pressing RESET zeroizes the time display.



3 Press DATA once again, and the fade out time display will flash, and enter the desired time using the dial.

4 Press SET to save the setting.

13-2. The Fade In Function

Before starting playback or recording, check to see that both the FADE IN and the FADE OUT indicators are lit on the display.

■ **Fading in play**

When starting playback, the sound will fade in over the preset time.

■ **Fading in record**

When starting record in ASSEMBLE mode, the input will fade in over the preset time.

13-3. The Fade Out Function

Both the FADE IN and the FADE OUT indicators must be lit on the display.

■ Fading out of play

Upon pressing STOP during play, the sound starts fading away and, at the end of the preset fade out time, the tape stops.

■ Fading out of record

Upon pressing STOP during record in ASSEMBLE mode, the input starts fading away and, at the end of the preset fade out time, the tape stops.

Monitoring the Fade In/Out Action

In PLAY you can monitor how the sound fades in and out at the digital and the analog outputs.

In RECORD also you can monitor the fade in/out action if the monitor is switched to TAPE.

SECTION 14 : PLAY AT VARIABLE SPEEDS

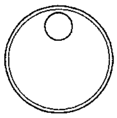
You can vary the play speed within the limits of +/-12.5 %, in 0.1 % steps.

- The play speed is variable only when the CLOCK switch is set to INT.

14-1. To Change the Pitch

1 During stop or play, press VARI SPEED to access a pitch menu.

The VARI SPEED LED will light up, and the "PITCH" indicator and the current pitch will show on the display.



2 Turn the dial to enter the desired amount of pitch. If the tape is playing, the pitch will change as you rotate the dial.

- To return to standard pitch, clear the pitch display to 00.0% with the dial or by pressing RESET.

14-2. Cancelling the Variable Speed Mode

Press the VARI SPEED key once again, and its LED will go out, and also the PITCH indicator.

- Cancelling the variable pitch mode does NOT erase the pitch changes from memory.

14-3. Checking the Current Pitch

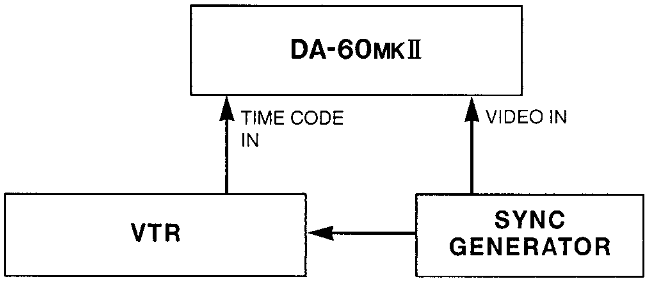
Once after having quitted the variable speed mode, if you want to check to see the current pitch, press MENUs until the PITCH indicator lights up on the display.

- You can change the pitch regardless of whether the VARI SPEED or the MENU keys are used to access the pitch menu.

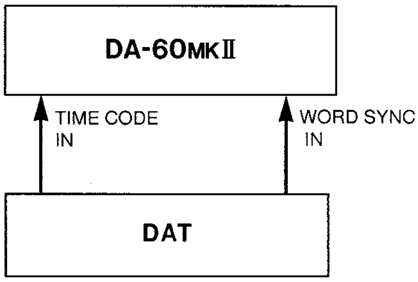
SECTION 15 : SLAVING TO EXTERNAL MACHINES

15-1. Hookup Examples

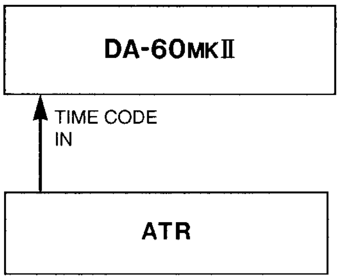
Slaving to a VTR



Slaving to a DAT



Slaving to an ATR



15-2. Setting-up the DA-60 MKII

Selecting a Reference Clock

Depending on your system hookup, set the CLOCK switch to :

- INT if the master is an ATR,
- EXT if the master is a DTR or
- VIDEO if the master is a VTR.

If EXT is selected, set the EXT SYNC switch to :

- WORD for referencing to the clock derived from the word sync in OR
- DIGITAL for referencing to the clock derived from the digital in (AES/EBU).

Normally, use the word clock. Only when it is not available, use the clock from the digital in.

Selecting a Playback Timecode Source

Refer to Section 8.

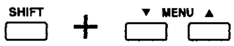
Selecting a Timecode Type

Refer to Section 7.

Checking the Timecode from the Master Machine

If you want to see the timecode numbers coming into the TIME CODE IN from the master machine, have the EXT TC indicator light up on the display by means of the MENU keys. You'll see the timecode numbers from the master on the right hand side of the display.

Checking the Timecode Type on the Tape in the DA-60 MKII



Hold SHIFT and press MENUs to access the menu "20 tPtc".

20 tPtc   30 ndF

To the right of the menu name shows one of the following indications :

- 30 ndF (30 f/s, non drop)
- 2997 ndF (29.97 f/s, non drop)
- 2997 dF (29.97 f/s, drop)
- 25 Ebu (25 f/s)
- 24 FiL (24 f/s)
- 30 dF (30 f/s, drop)

A broken line will show instead if no timecode is present on the tape.

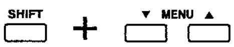
Selecting a Chase Mode

The DA-60 MKII offers two chase modes :

(1) Re-chase mode (default) — The DA-60 MKII duplicates every action of the master.

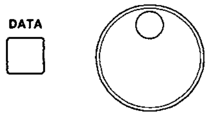
(2) Free mode — As soon as sync is achieved, the DA-60 MKII starts playing independently of the master.

To select the free mode :



1 Hold down SHIFT and repeatedly press MENU until the display shows :

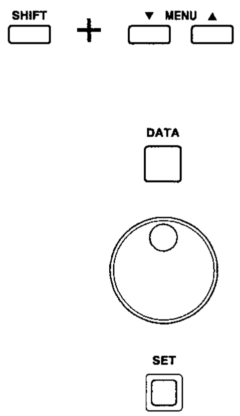
23 cHASE   rE-cHASE



2 Press DATA, and turn the rotary dial to change the blinking "rE-cHASE" to "FrEE".

3 Press SET to save the selection.

Timing the Timecode Output



The DA-60 MKII defaults to Analog mode so that the timecode output is timed to coincide with the analog output. If you intend to use the digital output, follow these steps, and the timecode output will be timed to coincide with the digital output.

1 Hold down SHIFT and repeatedly press MENU to access a timecode delay menu, which looks like this :

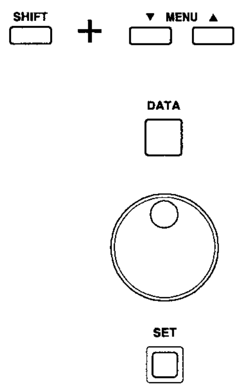
25 tcdLy   AnA

2 Press DATA, and the "AnA" will start blinking.

3 Turn the rotary dial to change the "AnA" to "diG."

4 Press SET.

Selecting a Play Start Point



When the DA-60 MKII chases the master and comes a certain distance short of it, the former starts playing at intermediate speeds, and as soon as sync is achieved, goes into normal play mode. If you want the DA-60 MKII to start playing only after locking to the master, follow these steps :

1 Hold down SHIFT and press MENU until the display reads :

27 cHS-A   PLAy

2 Press DATA. The "PLAy" will start blinking.

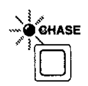
3 Turn the rotary dial to change the "PLAy" to "Loc".

4 Press SET.

15-3. Synchronization

We'll use the DA-60 MKII as a slave.

- If the inbuilt timecode generator is in Free run mode, no timecode is read from the tape. If in doubt, check the current setting at the menu "24 tcGEn" (pages 7 • 2 and 3).



1 Put the master machine into play mode.

2 Press CHASE on the DA-60 MKII.

- The CHASE LED is always lit regardless of whether it is chasing or syncing if the deck is in Re-chase mode. If the deck is in Free mode, the LED turns off as soon as the deck locks to the master.

- A SERVO LOCK indicator will light at the left of the display as soon as the deck locks to the master. But speaking of remembering this indicator lights also to simply show that the tape is correctly running.

- To disable the chase mode, press STOP.

Relative Difference Time Display

When you have the EXT TC indicator light with the MENU keys and timecode numbers from the master machine are showing on the right hand side of the display, press SHIFT, and a relative difference time is displayed to sub-frame accuracy (1/100 frame).

- The relative difference time is a time obtained by subtracting offset values from the absolute difference between the master and the slave timecode numbers.
- Pressing any other keys than SHIFT reverts the display to show the master timecode numbers.

15-4. Syncing with an Offset

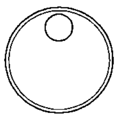
Offset can be programed at any time, even while syncing, up to +12:00:00:00.00, or down to -12:00:00:00.00, in 1 sub-frame (1/100 frame) steps.

Entering an Offset Value

1 Press MENU until "■ CHASE OFFSET" lights up on the display and below the indicator will show the current offset value.

2 Press DATA. The hour digits will start blinking.

Each time you press DATA, the next lower (right) two digits will blink.



3 When the digits you want to set are blinking, turn the rotary dial so as to display the desired number.

You can press RESET to clear the blinking digits to 00.

4 When you are satisfied with all the hour, minute, second, frame, and subframe displays, press SET.

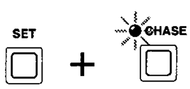
- If you enter or change the offset while the machines are syncing, you can hear the DA-60 MKII's sync point move relative to the master.

15-5. Auto Offset Entry

You can capture the current difference between the master and the slave timecode numbers "on the fly" so the DA-60 MKII gets synced up to a master with a lag corresponding to the offset value thus entered.

1 Let the master play, and also the DA-60 MKII (slave).

2 Press MENU until "■ CHASE OFFSET" lights up on the display and below the indicator will show the current offset value.



3 When the slave tape approaches the expected sync point, hold down SET and, when the point is reached, hit CHASE.

The time display now shows the captured offset time, the DA-60 MKII starting to chase the master.

- To disable the offset sync operation, press STOP.

15-6. Punch In and Out while Syncing

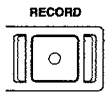
You can have the DA-60 MKII drop into and out of record while it is playing in sync with a master, manually or automatically.

> ***NOTE***
>
> *There must be at least 5 frames between the punch-in and out points.*

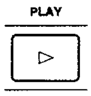
Manual Insertion

1 Enter the EDIT AUDIO mode.

2 Have the DA-60 MKII play in sync with a master.



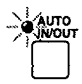
3 At the point where you want the DA-60 MKII to drop into record, hit RECORD.



4 Hit PLAY and the deck punches out of record.

Auto Insertion

1 Enter the EDIT AUDIO mode.



2 Press AUTO IN/OUT, and its LED will light.

3 Load a punch-in point into the MEMO 1 register, and a punch-out point into the MEMO 2 register.

4 Press CHASE.

5 After once sync is achieved, control the master transport to locate the DA-60 MKII to a point more than 5 seconds lower than the programed punch-in point.

When the DA-60 MKII plays up to the point 5 seconds lower the programed punch-in point, the AUTO IN/OUT LED will start blinking.

6 If a trial punch in (rehearsal) is required, press PLAY on the master.

7 When the trial punch in is over, press STOP on the master, and locate the DA-60 MKII to a point more than 5 seconds lower than the programed punch-in point by controlling the master transport.

15-7. Slaving to a Video Picture

The DA-60 MKII is or is not capable of getting synced up to rising edges of the video frame, as selected at a “26 SYnc P” menu.

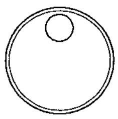
- The CLOCK select switch must be set to VIDEO and the Free mode must be selected at the menu "23 cHASE" (page 15 • 2).

To check or change the current selection at the menu “26 SYnc P” :

1 Hold down SHIFT and press MENU until the display shows :

26 SYnc P on

2 Press DATA. The "on" will start blinking.



3 Turn the rotary dial to change the "on" to "oFF" or vice versa.

4 Press SET.

SECTION 16 : OPERATIONS CONFORMING TO P2 PROTOCOL

16-1. Initial Settings

To control over a P2 interface the DA-60 MKII from editors/controllers which you'll connect to the RS-422 connector on the DA-60 MKII :

Make the following switch settings.
- REMOTE to ON
- CLOCK to VIDEO or WORD

If you set CLOCK to VIDEO, select "on" at the menu "26 SyncP".

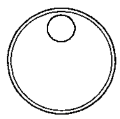
In addition, follow these steps :

To Get Ready to Access P2-related Menus

1 Hold SHIFT and press MENU until the display reads :

P2 cLoSE

2 Press DATA and the "cLoSE" will start flashing.



3 Turn the rotary dial to change the "cLoSE" to "oPEn".

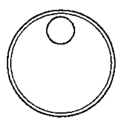
4 Press SET so you are ready to access P2-related menus.

Setting at a "9P-id" Menu

Follow these steps to select an ID with which the DA-60 MKII responds upon receiving a Device Type Request command from editors/controllers.

1 Hold SHIFT and press MENU until the display reads :

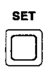
9P-id -0-7050



2 Press DATA and, as you turn the rotary dial, the following options will show in sequence :

- 0 - 7050	:PCM-7050
- 1 - 3000	:BVH-3000
- 2 - 950	:BVU-950
- 3 - 75	:BVW-75
- 4 - 9850	:VO-9850
- 5 - 2000	:BVH-2000
- 6 - 10	:DVR-10
- 7 - tEAC	:TEAC

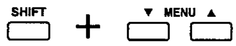
Editors/controllers use different procedures to control different machines. We recommend that you select the "PCM- 7050" setting. If your editor/controller does not identify the DA-60 MKII, try other settings.



3 Press SET to save the setting.

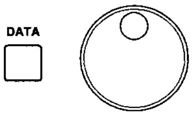
Setting at a "SPEEd" Menu

When the DA-60 MKII receives a REW or F FWD command from editors/controllers, the tape runs at 9 times the normal play speed or at 150 times, as selected at the menu "SPEEd". The default setting is 150. To change it to 9 :



1 Hold SHIFT and press MENU until the display reads :

SPEEd FASt 150



2 Press DATA and turn the rotary dial to change the "150" to "9".

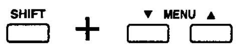
3 Press SET to save the setting.

- Depending on editors/controllers, they either only tell recorders to find specific points on tape (after receiving a locate command, recorders act independently to find the specified point themselves) or they continue to control recorders by issuing commands such as REW, F FWD, Shuttle, etc.

  In the latter case, fast winding the tape at 150 times the normal speed would result in a considerable overshoot ; autolocation may not complete at the expected point. To overcome this problem, select a 9 times speed at the menu "SPEEd".

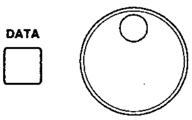
Setting at a "422AdrS" Menu

When controlling this unit from the TASCAM ES-61 edit controller by connecting to its ADDRESSABLE terminal, you have to select an address number as follows :



1 Hold SHIFT and press MENU until the display reads :

422AdrS 1



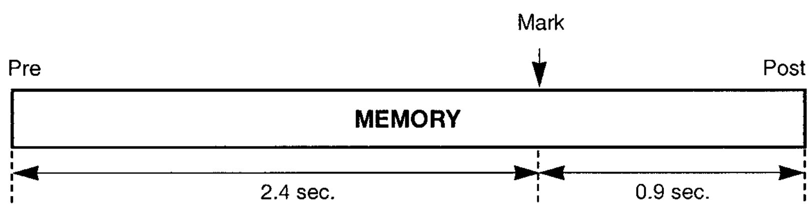
2 Press DATA, and turn the rotary dial to have the required number (1-16) be shown.

3 Press SET to save the setting.

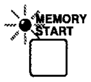
16-2. Memory Jog

This feature is used to fine tune marks to which you want to autolocate by pressing "GO TO" or an equivalent key on your editor/controller.

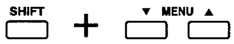
The fine tuning operation consists in playing seconds of audio preceding and following a marked point, to store them into a memory buffer for MEMORY START, and finding the exact edit point while auditioning the sound from the memory buffer.



Setting on the DA-60 MKII



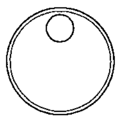
1 Enter MEMORY START mode by pressing the corresponding key.



2 Hold SHIFT and press MENU until the display reads :

JOG-Ed oFF

3 Press DATA and the "oFF" starts flashing.



4 Turn the rotary dial to change the "oFF" to "on".

5 Press SET to save the setting.

On the Editor/Controller

1 Mark an In or Out point.

2 Press GO TO, and the tape is located to the mark and audio is stored into memory. When storage is complete, the MEMORY START LED stops flashing, glowing solid.

3 Enter a Jog mode, and turn the Jog dial to hear the 'buffered' audio . As you rotate the dial to the left or right, the edit point will move back and forwards as confirmed by the timecode number display.

Each time you stop the dial, you will hear 150 msec. or 5 frames (30 msec.= 1 DAT frame) of audio before the trimmed point.

4 Enter a Shuttle mode. While the shuttle mode is 'stilled', turning the jog dial to the right offers a trial play, allowing you to hear 1 second of audio from the trimmed point on at the standard speed. The trial play will stop upon entering the shuttle still mode again.

- Turning the dial to the left has no effect at all.

5 When trimming the edit point to your satisfaction, replace the original point with the trimmed one.

- Audio before and after a marked point is saved also when activating PREVIEW on editors/controllers.

- The memory buffer in the DA-60 MKII is used for the Memory Jog function when the REMOTE switch is pressed on ; or for the Memory Start function if the switch is off.

SECTION 17 : LISTS OF MENUS

Menus are divided into two groups. Menus that you can access by pressing only MENU are grouped under "Group 1", and all other menus that you can access by holding SHIFT and pressing MENU are grouped under "Group 2".

Settings and changes at menus become the default (they are not lost when turning off power). See also SNAPSHOTS OF YOUR SETTINGS AT MENUS later in this section.

GROUP 1

Indication	Used for	See also p.
MEMO 1	Setting a point to which you want to autolocate or showing content of the corresponding register.	10 • 1
MEMO 2	(")	(")
FADE IN & FADE OUT	Setting/displaying the length of fade in and out. The fade in and out actions occur only when this menu is shown.	13 • 1 & 2
PITCH	Setting the pitch change or displaying the current setting.	14 • 1
(no indication)	Specifying a program for the deck to find it.	10 • 2
EXT TC	Displaying the timecode numbers coming into the TIME CODE IN.	15 • 2, 15 • 4
GEN TC	Displaying the timecode numbers as the inbuilt generator produces.	7 • 3
GEN TC SET	Setting the start time of the inbuilt generator.	7 • 3
CHASE OFFSET	Entering an offset or displaying the current offset value.	15 • 4
U-BIT EXT	Displaying U-BIT data coming from the exterior.	—
U-BIT	Displaying U-BIT data read from the tape loaded on the DA-60 MKII (if available).	—

GROUP 2

Menus of Group 2 can be accessed in either of two ways :
Press MENU while holding SHIFT, or
Rotate the DATA dial while holding SHIFT.
To switch the menu display (at the left) back to show timecode numbers, press either MENU key.

Menu	Option	Factory preset	Description	See also p.
1 Audio	cH-1, cH-2	cH-1	Selects a source for the MARGIN display.	17 • 3
2 Hour	XXXX (H)	—	Shows the head drum operating time.	17 • 3
3 PrEEP	oFF, on	oFF	Toggles Emphasis On/Off for the analog inputs.	5 • 4
4 coPY	00, 10, 11	00	Specifies a copy flag to be recorded.	5 • 2
5 At-id	-54,-60, -66 dB, oFF	-54	Specifies a level at the occurrence of which Start IDs are automatically marked during ASSEMBLE mode.	9 • 1
6 AtcuE	-54,-60,-66 dB	-54	Selects a trigger level of the auto cue circuit.	11 • 2
7 rEH-t	100 to 2500	2000	Sets the MEMORY START "rehearsal" time (in ms).	11 • 1
8 tALLy	StAndArd, rc-d6	StAndArd	Configures tally signals available at the parallel port for the optional RC-D6 remote or for other units.	17 • 3
9 FAdE	10, 50, 100	10	Specifies a time during which crossfade will occur at punch in and out points.	12 • 1
10 P-HLd	Auto, HoLd, oFF	Auto	Determines the level meter functions.	17 • 3
11 rLS-t	10, 50	10	Specifies a release time of the level meters.	—

(Cont.)

Menu	Option	Factory preset	Description	See also p.
12 iSdFt	oFF, iS on, iScuE	oFF	Determines whether only MEMORY START or a combination of MEMORY START and AUTO CUE is automatically activated each time the deck is turned on.	17 • 3
13 PrE-r	0 to 15	5	Selects a preroll time for insertion.	12 • 1
14 bAnd	nrr on, nrr oFF	nrr on	Sets the limit of variation in the frequency of external sync signals.	17 • 4
15 AtinP	oFF, on	oFF	Inputs to the DA-60 MKII can or cannot be monitored while in rewind, fast forward or stop, as selected at this menu.	17 • 4
-cALL-	FAct, rEG1, rEG2, rEG3	FAct	Reads menu settings from the corresponding registers.	17 • 5
-StorE-	rEG1, rEG2, rEG3	rEG1	Stores menu settings into one of the three registers.	17 • 5
-tc-	cLOSE, oPEn	cLoSE	Selecting "oPEn" allows you to access the following timecode-related menus.	7 • 1
20 tPtc	30ndF, 2997ndF, 2997dF, 25Ebu, 24FiL, 30dF	—	Shows the type of timecode available on the tape loaded on the DA-60 MKII.	15 • 2
21 Pbtc	Auto, tc, AbS	Auto	Depending on the setting at this menu, professional DAT timecode or ABS time data available on the tape is converted to SMPTE/EBU timecode numbers.	8 • 1
22 rEFtc	30ndF, 2997ndF, 2997dF, 25Ebu, 24FiL, 30dF	2997dF	Used to select a type of timecode to which the whole system is referenced.	7 • 2
23 cHASE	FrEE, rE-cHASE	rE-cHASE	The chase and lock action occurs only once or repeats, as selected at this menu.	15 • 2
24 tcGEn	oFF, rEc run, FrEE run, J-Sync	rEc run	The inbuilt generator operates depending on modes selected at this menu.	7 • 2
25 tcdLy	AnA, diG	AnA	Determines whether the timecode output is timed to coincide with the analog input or with the digital input.	15 • 3
26 SyncP	oFF, on	on	Video sync play is switched on or off from this menu.	15 • 6
27 cHS-A	PLAy, Loc	PLAy	The DA-60 MKII starts playing only when sync is achieved or when approaching a sync point, as selected at this menu.	15 • 3
--P2--	cLoSE, oPEn	cLoSE	Selecting "oPEn" allows you to access the following P2-related menus.	16 • 1
9P - id	-0- 7050, -1- 3000, -2- 950. -3- 75, -4- 9850, -5- 2000, -6- 10, -7- tEAC	-0- 7050	Selects an ID with which the DA-60 MKII responds upon the device type request from editors/controllres.	16 • 1
SPEEd	FASt 150, FASt 9	FASt 150	Determines whether the DA-60 MKII starts running at 150 times normal speed or at 9 times speed upon receipt of a rewind or fast-foward command from editors/controllers.	16 • 2
JoG - Ed	oFF, on	oFF	Upon receipt of locate commands such as "Cue up with data" from the editors/controller used, seconds of digital audio preceding and following a specific location can or cannot (depending on the setting as this menu) be stored into memory buffer inside the DA-60 MKII, to trim the location ("Memory Jog").	16 • 3
422AdrS	1 to 16	1	Used to select an address number when connecting to ADDRESSABLE terminal on the TASCAM ES-61 edit controller.	16 • 2

More Information about Some Menus of Group 2

• 1 Audio

When this menu is shown, the MARGIN display indicates how much level is available in the selected channel before clipping.

- This menu may be helpful for system calibration.
- When the MARGIN display reads -16 dB (reference value), the indicator ● lights. Smaller margins cause the ► indicator to light ; and larger margins cause the ◄ indicator to light.

• 2 Hour

When this indication shows on the right hand side of the display, the numbers you will see at the left show how many hours the head drum has spun since the initial utilization of the DA-60 MKII. This will be useful in carrying out your periodic maintenance plan on time.

• 8 tALLy :

At this menu you can select the destination of tally signals available at the ACCESSORY 1 port.

- Be sure to select "rc-d6" when connecting an optional remote (RC-D6) to the ACCESSORY 1 port.

• 10 P-HLd :

At this menu you can determine whether readings on the level meter will be held for a default time or as long as you want.

- When selecting "Auto", each peak reading will be held for about 1 second.
- When selecting "HoLd", readings will continue to be held until you select "Auto" or "oFF".
- When selecting "oFF", readings are not held.

• 12 iSdFt :

- When selecting "iS on", MEMORY START will automatically be activated upon powering up, as confirmed by the associated LED being lit.
- When selecting "iScuE", MEMORY START and AUTO CUE will both automatically be activated, as confirmed by both LEDs being lit.
- When you don't need those functions, select "oFF".

- **14 bAnd**

CLOCK MODE	nrr on	nrr oFF
EXT	Recording is possible when the frequency stability of the clock derived from the WORD or DIGITAL IN is within +/– 100 PPM.	Recording is possible when the clock frequency stability from the WORD or DIGITAL IN is within +/– 12.5%
VIDEO	DA-60 MKII is referenced to the clock derived from the VIDEO IN. The VIDEO indicator will flash and the internal clock mode will be entered if the selected TC format does not match with the incoming video signal frequency.	Deferences between the selected TC format and the incoming video signal frequency don't disable the video clock mode.

When selecting "nrr oFF", the DA-60 MKII can record at the pitch variation of +/–0.1 % despite the inconsistency between the timecode rate and the video input frequency (as shown in tables below). When referencing to the internal clock, the deck allows recording at variable speeds within the limits +/–0.2 %.

Setting for Pull Up (Recording at +0.1% speed for Film production)

	Without 30 Hz Video Sig.	With 30 Hz Video Sig.
CLOCK SW	INT	VIDEO
14 bAND	-----	nrr oFF
22 rEFtc	29.97 ndF / 29.97 dF	29.97 ndF / 29.97 dF
VIDEO IN Sig.	-----	30 Hz
Pitch Change	+0.1% (VARI SPEED)	-----

Setting for Pull Down (Recording at –0.1% speed for HD TV)

CLOCK SW	VIDEO
14 bAND	nrr oFF
22 rEFtc	30 ndF / 30 dF
VIDEO IN Sig.	29.97 Hz

- **15 AtinP**

When selecting "on" at this Auto inPut menu, the monitor automatically switches to the input ("SOURCE") whenever you press REW, F FWD, STOP, LOC or ID SEARCH while in ASSEMBLE mode.

- Once after the monitor is switched to the input, the MONITOR switch can be used to toggle between input/tape options. The broken line (---) in the table indicates that the monitor can be switched only with the MONITOR switch.

REC FUNCTION			During RECORD	During PLAY	STOP REW F FWD
ASSEMBLE	15 AtinP	oFF	—	Tape	—
		on	Tape	Tape	Source
EDIT AUDIO			Source	Tape	Tape

"SNAPSHOTS" OF YOUR SETTINGS AT MENUS

You can take three snapshots of all your settings at menus (of Groups 1 and 2), and switch the DA-60 MKII from snapshot to snapshot.

As you make settings at menus, the DA-60 MKII stores your settings into register 0. To take a snapshot of the contents of this register and save to another register for later recall :

1. Hold SHIFT and press MENU until "-storE-" shows on the left hand side of the display. At the right you'll see a register indicator be flashing.

2. Press DATA, and select one of the three registers with the rotary dial. Upon pressing SET the contents of register 0 are stored into the selected register.

   • Contents of register 0 are not lost when switching power off, but they are replaced when recalling a snapshot from other registers as follows :

To recall a snapshot

1. Hold SHIFT and press MENU until "-cALL-" shows on the left hand side of the display.

2. Press DATA and turn the rotary dial to have the desired register number appear at the right. Upon pressing SET the contents of the selected register are read into register 0.

   • If you select "FActory", all the menus will be switched back to their factory presets.

SECTION 18 : ERROR MESSAGES EXPLAINED

18-1. Coded Error Messages

When an error condition exists inside the DA-60 MKII, this generates such messages as shown here.

Servo-related Error		
Error Code	Problem	Remedy
1-1	Incorrect data transmission to/from the Control Circuit Board	A
1-2	No rotation of drum motor	A
1-3	Cassette not loaded correctly	A
1-4	No rotation of takeup reel motor	A
1-5	No rotation of supply reel motor	A
1-6	Condensation on the head drum	B

Digital-related Error		
Error Code	Problem	Remedy
2-1	Incorrect data transmission to/from the Control Circuit Board	A
2-2	Incorrect function of system clock circuits	A

Control-related Error		
Error Code	Problem	Remedy
3-1	Backed-up data destructed	C

RAM-related Error		
Error Code	Problem	Remedy
4-1	Incorrect data transmission to/from the Control Circuit Board	A

Sync-related Error		
Error Code	Problem	Remedy
5-1	Incorrect data transmission to/from the Control Circuit Board	A

Remedy :

A. Switch the power off, then switch it on again.
B. Leave the unit turned on for 1 hour or 2 until the error message gose out.
C. There is no detriment effect on the general functions of the unit, but no data about your settings can be retained in a backup memory when the unit is turned off unless lithium batteries are replaced.

If you use remedy A or B and error messages don't turn off, or if the lithium batteries need to be replaced, please contact TASCAM at the address shown on the back of the DA-60 MKII's manual or your nearest TASCAM dealer.

18-2. Flashing Messages

Wrong settings and connections would cause the following messages to appear.

Indication Flashing	Problem and Remedy
WORD	• WORD EXT SYNC is selected, but the nesessary clock is not coming in, and the unit is referenced to the internal clock instead. Check to see if a word sync signal is plugged into the word sync in. If it is OK, check if any different sampling rate data than the incoming one is already recorded on the tape (see also "Fs" below).
DIGITAL	• DIGITAL EXT SYNC is selected, but the DA-60 MKII is referenced to the internal clock. Check to see if a digital audio is plugged into the digital in. If it is OK, check if any different sampling rate data than the incoming one is already recorded on the tape (see also "Fs" below). • The clock selected on the DA-60 MKII is not the same as the incoming clock. Select the correct clock or change the clock setting on the external machine to match the clock selected on the DA-60 MKII
VIDEO	• VIDEO CLOCK is selected, but the DA-60 MKII is referenced to the internal clock. Check to see if video signal is plugged into the video in, or if the frame rate (of time code) selected on the DA-60 MKII matches the incoming frame rate.
EXT TC	• EXT TC is selected to record on the DA-60 MKII, but no time code is coming in. Check to see if time code is plugged into the time code in. This indicator flashes also when the DA-60 MKII is in Chase mode but no timecode is coming from a master transport (because this is stopped or for any other reasons).
Fs 48.0 k	• The tape inserted to the DA-60 MKII carries some material already recorded at 44.1 kHz and you attempted to record additional material to augment the original one, but the Fs switch is set to 48.0 kHz or the incoming rate is 48.0 kHz. If the Fs switch is set to 48.0 kHz, set it to 44.1 kHz. If the incoming rate is not 44.1 kHz, don't use that source. It is not recorded correctly no matter how the Fs switch is set.
Fs 44.1 k	• Similar to the above case, but the old material was recorded at 48.0 kHz, and the Fs switch is set to 44.1 kHz or the incoming rate is 44.1 kHz.

18-3. Other Messages

If you tried invalid operations, the following messages will appear.

Indication	Problem
- iLLEGAL -	You tried to activate a function which can not operate in the current mode of operation (e.g., you pressed AUTO IN/OUT while the REC FUNCTION LED is off).
- rEc ProtEct -	You pressed REC FUNCTION while the tape is write-protected.
- rEc Function -	You tried to record or pressed ID SELECT while the REC FUNCTION LED is off.
- no tc in -	You tried to enter record mode without plugging in any external timecode while the inbuilt timecode generator is off.
- d-in Error -	You tried to record without plugging any source into the digital input while the INPUT select switch is set to DIGITAL.
- bot -	You attempted to let the tape run in reverse by pressing REW etc., while it is at the BOT (Beginning Of Tape).
- Eot -	You attempted to let the tape run in the foward direction by pressing PLAY, F.FWD etc., while it is at the EOT (End Of Tape).
- not LocAl -	Controls are pressed on the DA-60 MKII while REMOTE is pressed on.

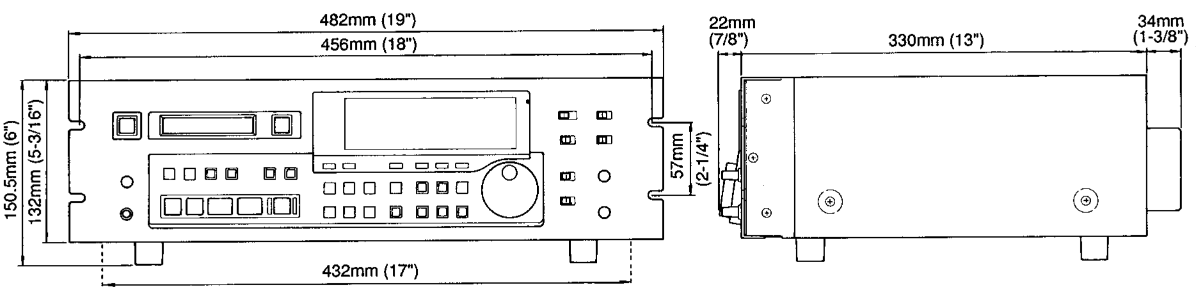
SECTION 19 : SPECIFICATIONS

Type : Rotary head digital audio tape recorder
Tape speed : 8.15 mm/sec. (12.225 mm/sec. supported)
Recording/play time : 120 minutes (with 120-minute tape)
Fast-winding time : 60 seconds or less (approx.)(with 120- minute tape)
Error correction : Duplex Reed Solomon code
Channel : 2-channel stereo
Quantization : 16-bit linear
Sampling rate : 44.1 kHz (recording/playback)
48 kHz (recording/playback)
Emphasis : 50 µsec./15 µsec.
Frequency response (recording/playback) :
5 Hz to 22,000 Hz +/-0.5 dB
Signal-to-noise ratio : Better than 94 dB (emphasis Off)
Better than 98 dB (emphasis On)
Dynamic range : Better than 94 dB (emphasis Off)
Better than 98 dB (emphasis On)
Total harmonic distortion :
Less than 0.004 % (recording/playback overall, at 1 kHz, at full-scale reading)
Channel separation : Better than 90 dB (at 1 kHz)
Wow and flutter : Unmeasurable (less than +/-0.001 %)

Analog I/O
Line In : XLR-type connector (XLR-3-31) x2 (pin 2 Hot)
Nominal input level : +4 dBm, bal.
Input impedance : 20 kohms
Line Out : XLR-type connector (XLR-3-32) x2 (pin 2 Hot)
Nominal output level : + 4 dBm, bal.
Maximum output level : +20 dBm, bal.
Output impedance : Less than 10 ohms
Monitor Out : RCA jack x2
Nominal output level : -10 dBV, unbal.
Output impedance : Less than 610 ohms
Headphones Out : 1/4" jack x1
Output power : 100 mW + 100 mW (into 8 ohms)
Digital I/O
Input : XLR-type connector (XLR-3-31) x1
Format : IEC958 Type I (AES/EBU)/Type II (SPDIF), auto selection
Output : XLR-type connector (XLR-3-32) x1
Format : IEC958 Type I (AES/EBU)

Timecode I/O
Input : XLR-type connector (XLR-3-31) x1
Nominal input level : 2Vp-p, bal.
Input impedance : 10 kohms
Output : XLR-type connector (XLR-3-32) x1
Nominal output level : 2Vp-p, bal.
Output impedance : 75 ohms
Word sync signal
Input : BNC connector x1
Nominal input level : Equivalent to TTL, unbal.
Input impedance : 75 ohms
Output : BNC connector x1
Nominal output level : Equivalent to TTL, unbal.
Output impedance : 75 ohms
Thru : BNC connector x1
Nominal output level : Equivalent to TTL, unbal.
Output impedance : 75 ohms
Video sync signal
Input : BNC connector x1
Nominal input level : 1 Vp-p, unbal.
Input impedance : 75 ohms
Control interface
ACCESSORY 1 : D-sub 37-pin connector (parallel) x1
Input/output level : Equivalent to TTL
RS-422 : D-sub 9-pin connector (serial) x1
Power requirements :
USA/CANADA : 120 V AC, 60 Hz
U.K./EUROPE : 230 V AC, 50 Hz
AUSTRALIA : 240 V AC, 50 Hz
General Export Model : 120/230/240 V AC, 50/60 Hz switchable
Power consumption : 58 Watts
Dimensions (W x H x D) : See illustration below.
Weight : 11.5 kg (25-6/16 lbs)

- *In these specifications, 0 dBV is referenced to 1 Volt, and 0 dBm is referenced to 0.775 Vrms.*
- *Changes in specifications and features may be made without notice or obligation.*



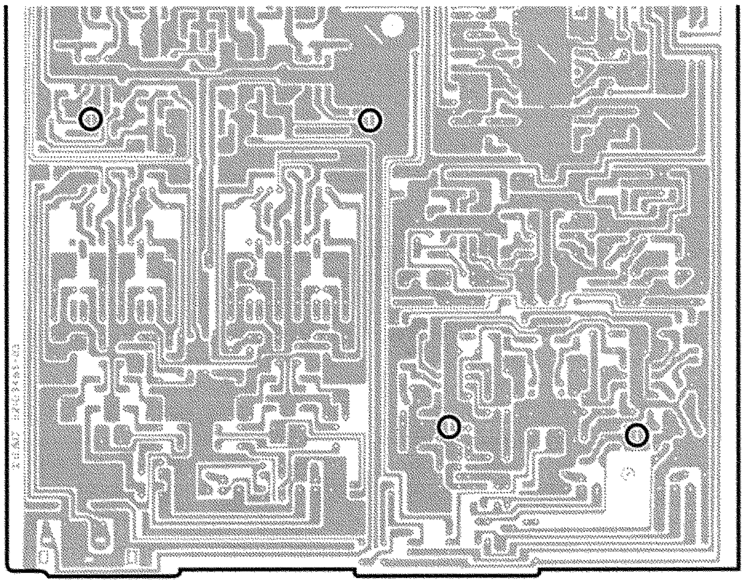
SECTION 20 : OPTIONAL FEATURES OF REFERENCE LEVEL

> ***WARNING***
>
> *Ask the TASCAM service technician to perform these modifications. Performing these or any other mods yourself places the Warranty in jeopardy.*

SMPTE Requirement

For a +4 dBu input signal to cause the level meter to read –20 dB instead of –16 dB as in the original, solder-short-circuit at four points indicated below. Maximum level at the analog output changes from +20 dBu to +24 dBu.

AUDIO PCB Assy



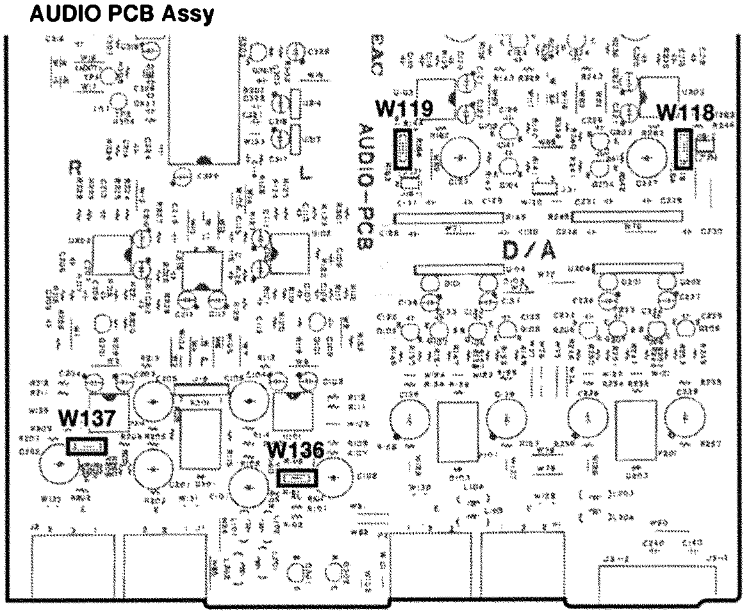
O : Short-circuit here

EBU Requirement

A +6 dBu input signal causes the level meter to read –9 dB. Maximum level at the analog output changes to +15 dBu.

Cut four jumper wires indicated below.

AUDIO PCB Assy



▭ : Cut here

TASCAM
TEAC Professional Division

DA-60MKII

TEAC CORPORATION	3-7-3, Nakacho, Musashino-shi, Tokyo 180, Japan Phone: (0422) 52-5081
TEAC AMERICA, INC.	7733 Telegraph Road, Montebello, California 90640 Phone: (213) 726-0303
TEAC CANADA LTD.	340 Brunel Road, Mississauga, Ontario L4Z 2C2, Canada Phone: 905-890-8008
TEAC UK LIMITED	5 Marlin House, Marlins Meadow, The Croxley Centre, Watford, Herts. WD1 8YA, U.K. Phone: 01923-819699
TEAC DEUTSCHLAND GmbH	Bahnstrasse 12, 65205 Wiesbaden-Erbenheim, Germany Phone: 0611-71580
TEAC FRANCE S.A.	17, Rue Alexis-de-Tocqueville, CE 005 92182 Antony Cedex, France Phone: (1) 42.37.01.02
TEAC NEDERLAND BV	Perkinsbaan 11, 3439 ND Nieuwegein, Nederland Phone: 03-402-30229
TEAC AUSTRALIA PTY., LTD. A.C.N. 005 408 462	106 Bay Street, Port Melbourne, Victoria 3207, Australia Phone: (03) 9644-2442
TEAC ITALIANA S.p.A.	Via C. Cantù 5, 20092 Cinisello Balsamo, Milano, Italy Phone: 02-66010500

PRINTED IN JAPAN 0297U0.5 M-1198C

TEAC TECHNICAL INFORMATION

9614

TASCAM DA-60/SY-D6, Upgrading

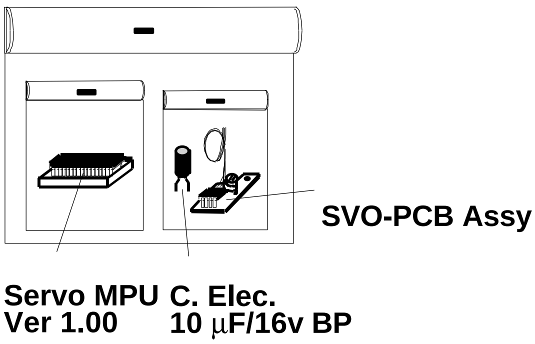
7th June 1996

Upgrading of DA-60/SY-D6 system is now ready.
"Upgrade" here means that DA-60/SY-D6 can have all features of DA-60Mk2.
Difference is only on the AD and DA section.

Following three parts are all for the upgrade.

1	P/No. V000544-00A	Servo MPU Update Kit, DA-60 which consists of: SVO PCB Assy Servo MPU Version 1.00 Capacitor, Elec., 10 µF/16V BP
2	P/No. S001981-00B	EP ROM, SYSTEM Version 6.01, DA-60Mk2
3	P/No. S001983-00A	EP ROM, SYNC Version 3.00, DA-60Mk2

V000544-00A Servo MPU Update Kit, DA-60

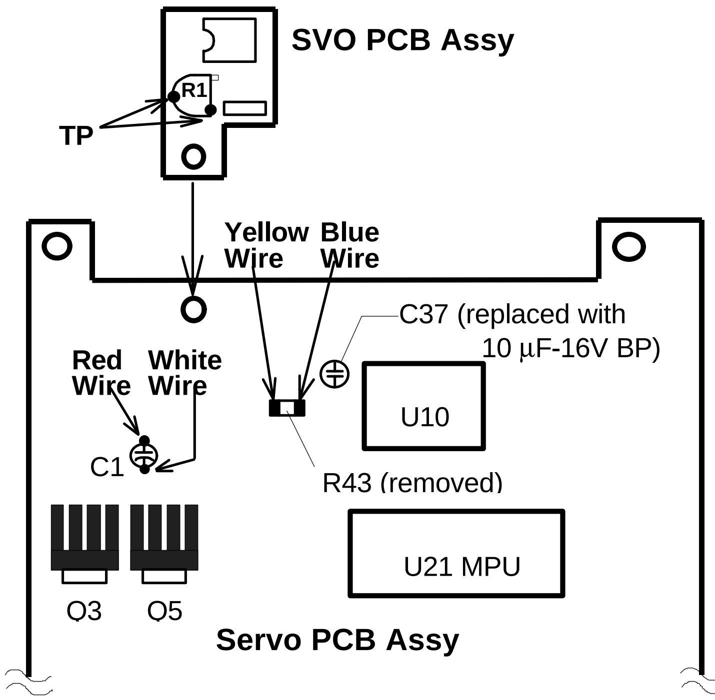


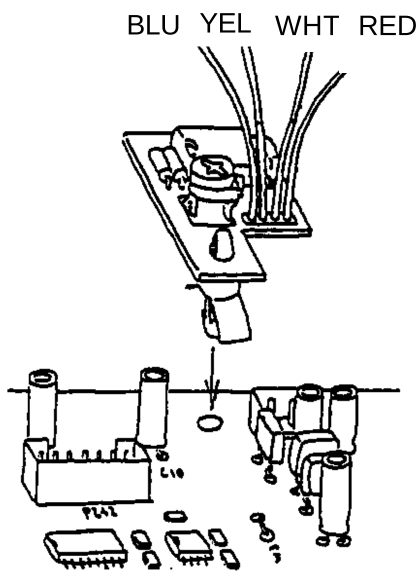
The SVO PCB Assy is newly developed to align capstan offset voltage out of U10-37 (HD49228FS HITACHI Servo LSI) which varies in batch to batch. By this aligned voltage, more delicate control of the capstan becomes possible through the Servo MPU version 1.00. The features explained hereinafter become to be realized therefore.

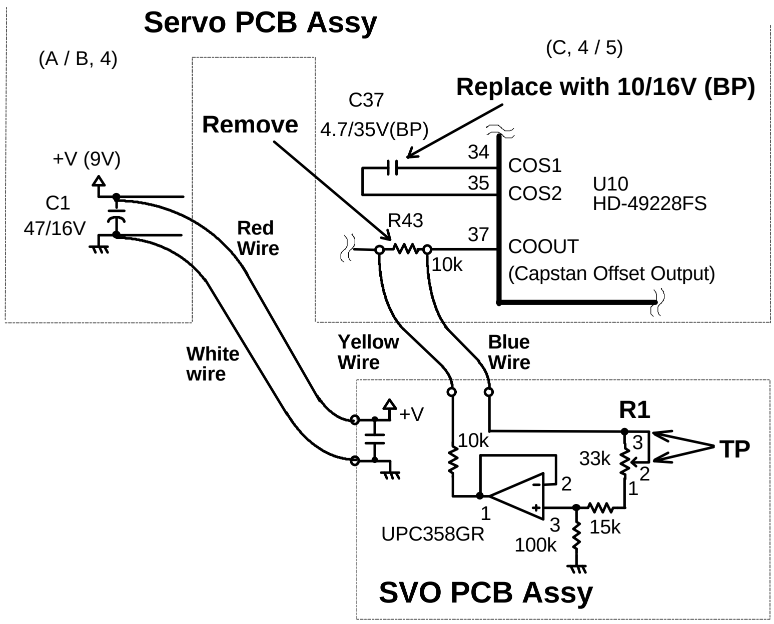
See page 2 for upgrading procedure.
See page 3 and 4 for the features which becomes available by this upgrading.

DA-60, Upgrading Procedure

1 On the Control PCB Assy, replace SYSTEM ROM to Version 6.01.
2 On the Sync PCB Assy (SY-D6), Replace SYNC ROM to Version 3.00.
3 Install the Servo MPU Update Kit onto the Servo PCB Assy as following:
3-1 Replace Servo MPU with Version 1.00.
3-2 Remove R43.
3-3 Replace C37 with 10µF/16V BP.
3-4 Mount and wire the SVO PCB Assy.
3-5 Make Assemble Recording with 120 minutes tape in the vicinity of EOT (End Of Tape) area.
3-6 During recording, observe **TP** with an oscilloscope. Adjust **R1** on the SVO PCB Assy so that center of the waveform comes to +2.7 ~ 2.8 V.







DA-60, Features on Upgrading

1 TC Jam-Sync

Continuous TC recording is possible by selection of Generator mode.

[24 tcGEn J-Sync **]**

In case no TC has been striped, start REC with [rEc run]

2 Immediate Lock-up

After the DA-60 is located to Preroll point, it goes into STILL mode (pinch roller engaged) instead of STANDBY mode. This can much shorten the lock-up time.

3 Further P2 Protocol Supported

Edit commands from some of Non-Linear Video Editing systems through 9 pin control can be accepted.

4 Punch Out Operation under Assemble Recording

Punch Out can be executed by pressing PLAY key.

5 Short Cut Operation

The following selecting menu and data input can be directly operated by double depression of keys:

a AUTO ID mode, ON / OFF

Press [SHIFT] and [ID SELECT], then ON/OFF is flip changed. Previous AUTO ID settled level is held when OFF to ON is made.

b PGM Number Search

Press [SHIFT] and [LOC1] to enter PGM Number Search menu and lower two digits of Locate P No blink to wait DIAL input. Turn DIAL for desired P No and press [LOC1] to execute searching.

c TC Generator Mode

Press [SHIFT] and [MARGIN] to enter **[** 24 tc GEn XXXX **]** menu display.

6 Deletion/Addition/Change of Menu

Menu 2 can be selected by pressing [SHIFT] and turning DIAL clockwise or counter-clockwise while previous approach of pressing [SHIFT] and [MENU] is also valid.

[Loc-P **]** menu in **[** -- P-- **]** group is deleted.

[422AdrS **]** menu in **[** -- P2 -- **]** group is added. And Address No of ' 1 ' ~ ' 16 ' can be selected when DA-60 is connected to ADDRESSABLE terminal of the ES-61.

[J-Sync] is added in **[** 24 tcGEn **]** menu for TC Regenerate REC.
Total selection is therefore changed from 3 to 4 as :

[oFF], [rEc run], [FrEE run], [J-Sync]

Preroll time set of seconds in「 13 PrE-r 」menu is expanded :

[5] ~ [9] ⇒ [0] ~ [15]

This time set is to be renewal when Preroll time is commanded from Editors through 9 Pin Control.

7 Supporting the Branches of " iLLEGAL" Display

[rEc ProtEct **]** appears when REC is attempted on DAT Cassette of which the write protect window is opened.

[rEc Function **]** appears when REC is attempted or ID SELECT key is operated while REC FUNCTION LED is not selected to light.

[no tc in **]** appears when REC is attempted without feeding external TC while EXT TC is selected.

[d-in Error **]** appears when REC is attempted without feeding Digital input while DIGITAL is selected.

TEAC TECHNICAL INFORMATION

TASCAM DA-60(MK2), ROM Upgrade (SYS V6.05)

No. **9831**
DATE 22nd July 1998

SYSTEM ROM, U21 on the Control PCB Assy has been upgraded on the products with S/No. 0100001 and higher.

SYS V6.05, P/No. S001981-00C

Problem corrected by SYS V6.05

Punch in/out points had lagged by about 80 µsec. This causes DC component and it could be heard as a click noise. This symptom will occur only when punch in/out is made with a single sine wave.

Notice: Music signal is not affected as it is made of harmonic tones.

Additional Features

Refer to attached sheets for the details.

DA-60MK2 (Version 6.05 and higher)

UPDATE

The following new functions are available in addition to those described in the owner's manual:

Single Play Mode

This mode allows you to play just one program, stopping automatically at the beginning of the next program.

1 Press MENU to display "Single-P" in the time readout display (right).
2 When this indicator is displayed in play mode, the deck automatically stops if it detects the next start ID.

- In Memory start mode (MEMORY START LED lights), the deck stores about the first four seconds of playback sound from the beginning of start ID into a memory buffer and sets the deck to Memory Start Standby mode.

■ Press MENU to cancel single play mode.
■ By short-circuiting pin-26 of Accessory 1 output to GND, the deck can be set to single play mode. Open pin-26 to cancel single play mode.

Recording/Erasing Skip IDs

Recording Skip IDs

1 Set the record mode to EDIT SUB (edit subcode).
2 Press ID SELECT to display "WRITE" in display window and "short-id" in the time readout display (right).
3 Press EXE at the desired point to record a skip ID during play. The indicator "WRITE" blinks on the display and the deck records the skip ID for about one second.

■ In order to record the skip ID correctly, proceeds as follows.
(1) In play mode, press MEMO 1(2) at the point where you want to record the skip ID.
(2) Follow the "MEMORY START OPERATIONS" (p.11.1) procedure to store audio into a memory buffer, to audition the "buffered" audio, or to trim your memory start point.
(3) After pressing SET to change the start point, set the record mode to "EDIT SUB".
(4) Press ID SELECT to display "WRITE" in the display window and "Short-id" in the time readout display(right).
(5) Press EXE to execute recording of the skip ID from that point.

Erasing Skip IDs

1 Set the record mode to EDIT SUB (edit subcode).
2 Press ID SELECT to display "ERASE" in the display window and the "Short-id" in the time readout display (right).
3 Press EXE in stop mode to rewind the tape to the previous skip ID. The deck erases the skip ID while playing the tape.

■ If the deck detects a skip ID while the "Short-id" is displayed in the time readout display(right), "Short-id 1" lights up in the display.

Skip Mode

When a skip ID is detected during play, this mode fast-forwards the tape to the next start ID and stops automatically just before the start point of the start ID.
● If the deck is in Memory Start mode(MEMORY START LED lights), the deck stores about the first four seconds of playback sound from the beginning of the start ID into a memory buffer and sets the deck to Memory Start Standby mode.

1 Hold down SHIFT and press MENU to show the following menu on the time readout display of the display window.
● This setting is added after "422AdrS" in the P2 section of menu 2.

"ShortEninG oFF"

2 Press DATA, and "oFF" will start blinking. Turn the rotary dial to change it to "on", then press SET.

TEAC TECHNICAL INFORMATION

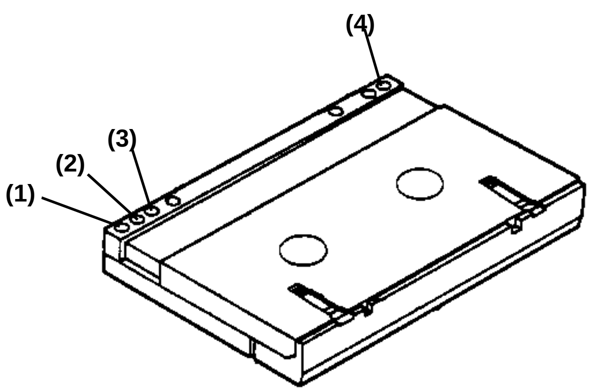
9807
6th March 1998

TASCAM DA-60(MK2), Clearing Drum Time

Strictly Confidential

Drum time can be cleared by the following procedure.

1 Open a hole (4) on a cassette.
2 Set the write protect switch to close (recording enable).
3 Wind a tape to EOT (End of Tape).
4 Press STANDBY then STANDBY LED turns off.
5 Access to [1 Audio] in the MENU 2.
6 Hold SHIFT then press STANDBY.



TEAC TECHNICAL INFORMATION

9625

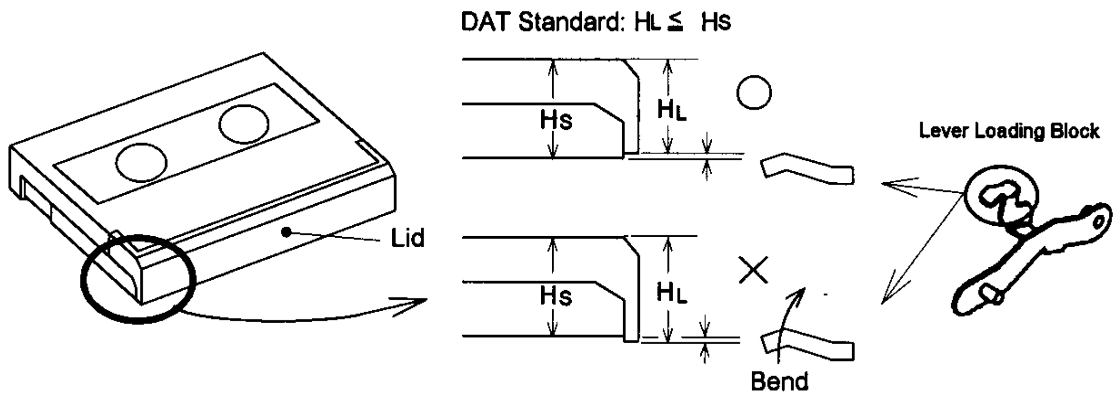
TASCAM DA-60/60MK2, Lever Loading Block

4th October 1996

Problem:
Cassette cannot be loaded. If the cassette is inserted strongly, Lever Loading Block which opens the lid of the cassette may be bent down.

Cause:
Accumulation of tolerance being met with the height of lid and the Lever Loading Block.
Height of lid is specified with the DAT standard as HL should be shorter than HS as shown below.
However, some brands of cassette have a longer lid.
Also Lever Loading Block is fixed with slight play in its shaft.

Solution:
Replace the Lever Loading Block, **P/No. 57618697-00** if a bend is found, Replace an original 0.25t washer fixing the Lever Loading Block with 0.35t, **P/No. 57618633-00**. This can reduce play on the Lever Loading Block. DA-60MK2 S/No, 40001 and higher have this 0.35t Lock
Washer.
Refer to next page which explains actual work.

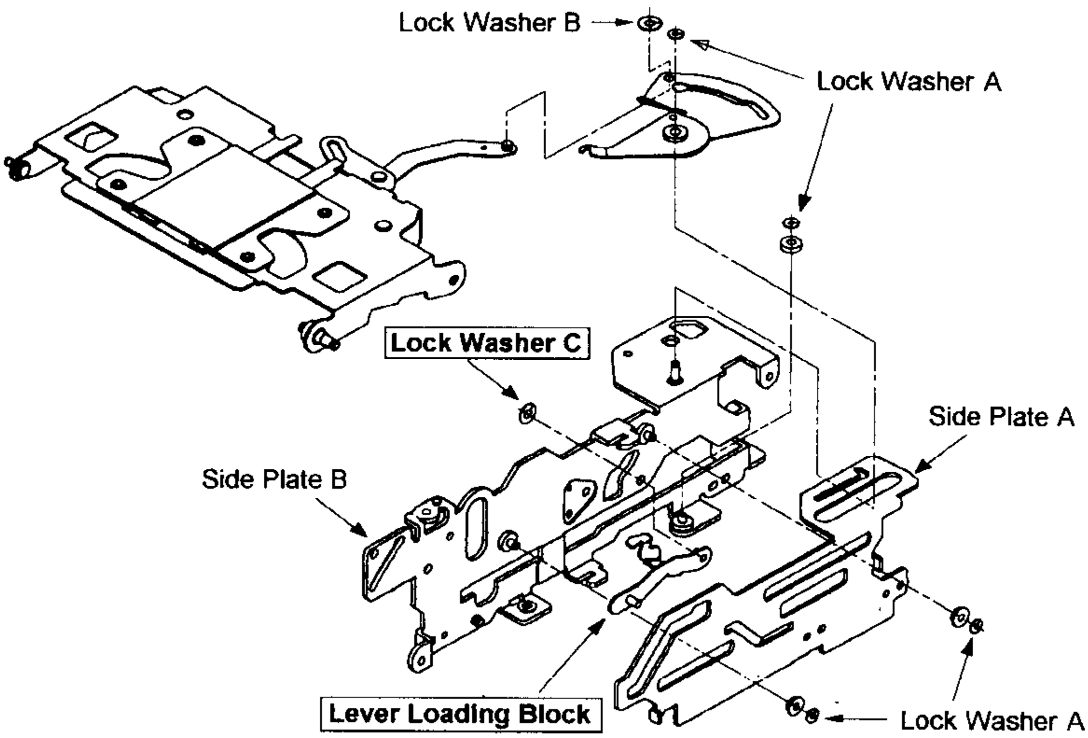


Disassembling

I Remove Lock Washer A (4 pcs) and Lock Washer B then Slde Plate A can be removed.
2 Remove Lock Washer C then Lever Loading Block can be removed from Side Plate B.
3 Replace the Lever Loading Block if a bend isfooud.

Assembling

1 Fix the Lever Loading Block with new Lock Washer (0.35 t).
2 Assemble all parts with new Lock Washers. Do not use Lock Washers once removed.
3 After assembling, check if the lid does not hit the Lever Loading Block.



Lever Loading Block	P/No 57618697-00
Lock Washer A	P/No 57618621-00
Lock Washer B	P/No 57618642-00
Lock Washer C (0.35t)	P/No.57618633-00 * Do not use the original 0.25t.

Semiconductor Corporation

CS5336 CS5338 CS5339

16-Bit, Stereo A/D Converters for Digital Audio

Features

- Complete CMOS Stereo A/D System
  - Delta-Sigma A/D Converters
  - Digital Anti-Alias Filtering
  - S/H Circuitry and Voltage Reference
- Adjustable System Sampling Rates including 32kHz, 44.1 kHz & 48kHz
- Low Noise and Distortion
  - >90 dB S/(N+D)
- Internal 64X Oversampling
- Linear Phase Digital Anti-Alias Filtering
  - 0.01dB Passband Ripple
  - 80dB Stopband Rejection
- Low Power Dissipation: 400 mW
  - Power-Down Mode for Portable Applications
- Evaluation Board Available

General Description

The CS5336, CS5338 & CS5339 are complete analog-to-digital converters for stereo digital audio systems. They perform sampling, analog-to-digital conversion and anti-aliasing filtering, generating 16-bit values for both left and right inputs in serial form. The output word rate can be up to 50 kHz per channel.

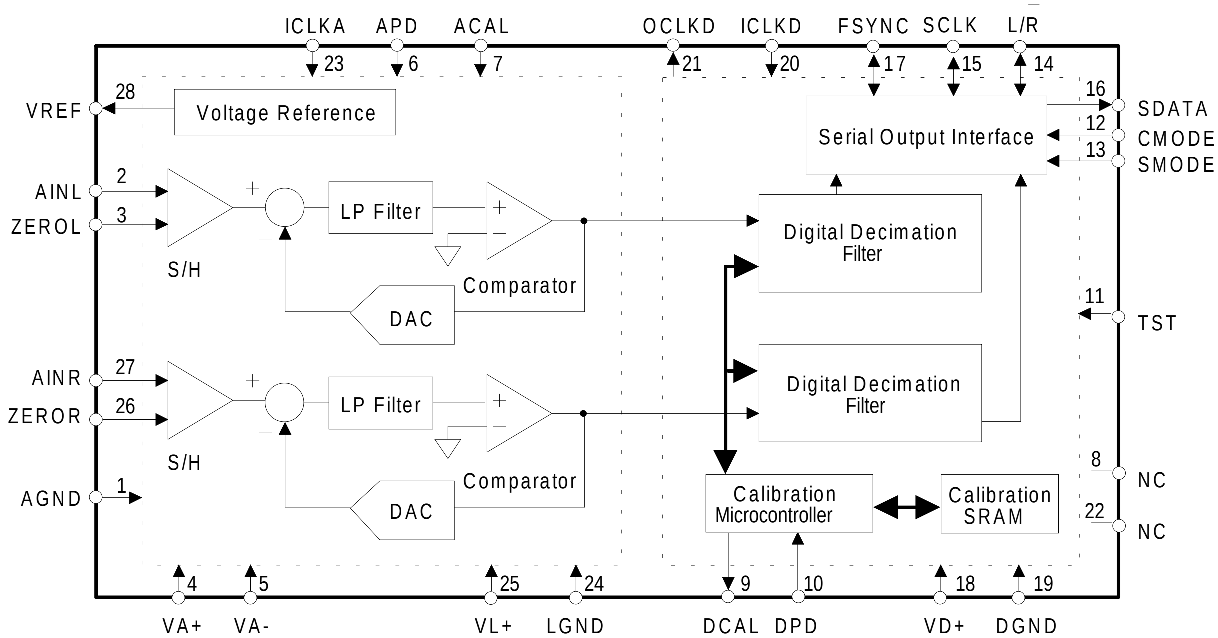
The ADCs use delta-sigma modulation with 64X over-sampling, followed by digital filtering and decimation, which removes the need for an external anti-alias filter.

The CS5336 & CS5338 have an SCLK which clocks out data on rising edges. The CS5339 has an SCLK which clocks out data on falling edges.

The CS5336 has a filter passband of dc to 22kHz. The CS5338 & CS5339 have a filter passband of dc to 24 kHz. The filters have linear phase, 0.01 dB passband ripple, and >80 dB stopband rejection.

The ADC's are housed in a 0.6" wide 28-pin plastic DIP, and also in a 0.3" wide 28-pin SOIC surface mount package. Extended temperature range versions of the CS5336 are also available.

ORDERING INFORMATION: See Page 3-59



Crystal Semiconductor Corporation
P.O. Box 17847, Austin, TX 78760
(512) 445-7222 FAX: (512) 445-7581

ANALOG CHARACTERISTICS

(Logic 0 = GND; Logic 1 = VD+; K grade: T_A = 25°C; B and T grades: T_A = T_{MIN} to T_{MAX}; VA+, VL+,VD+ = 5V; VA- = -5V; Full-Scale Input Sinewave, 1kHz; Output word rate = 48 kHz; SCLK = 3.072 MHz; Source Impedance = 50Ω with 10 nF to AGND; Measurement Bandwidth is 10 Hz to 20 kHz; unless otherwise specified.)

Parameter	Symbol	CS5336,8,9-K			CS5336-B			CS5336-T			Units
		Min	Typ	Max	Min	Typ	Max	Min	Typ	Max	
Specified Temperature Range	T_A	0	to	70	-40	to	+85	-55	to	+125	°C
Resolution		16	-	-	16	-	-	16	-	-	Bits
Dynamic Performance											
Dynamic Range		92.7	95.7	-	90	93.5	-	84	92	-	dB
Signal-to-(Noise + Distortion); THD+N	S/(N+D)	90.7	92.7	-	85	89	-	82	86	-	dB
Signal to Peak Noise		-	96	-	-	95	-	-	94	-	dB
Total Harmonic Distortion	THD	.0025	.001	-	.005	.001	-	.013	.005	-	%
Interchannel Phase Deviation		-	.0001	-	-	.0001	-	-	.0001	-	°
Interchannel Isolation (dc to 20 kHz)		100	106	-	90	106	-	83	96	-	dB
dc Accuracy											
Interchannel Gain Mismatch		-	0.01	0.05	-	.01	.05	-	.01	0.1	dB
Gain Error (includes Vref tolerance)		-	±1	±5	-	±2	±5	-	±3	±6	%
Gain Drift (includes Vref drift, Note 1)		-	25	-	-	70	-	-	70	-	ppm/°C
Bipolar Offset Error (Note 2)		-	±5	±15	-	±10	±30	-	±16	±65	LSB
Offset Drift (Note1)		-	15	-	-	20	-	-	20	-	ppm/°C
Analog Input											
Input Voltage Range (±Full Scale)	VIN	±3.5	±3.68	-	±-3.5	±3.68	-	±3.5	±3.68	-	V
Input Impedance	ZIN	-	65	-	-	65	-	-	65	-	kΩ
Power Supplies											
Power Supply Current (VA+)+(VL+) with APD, DPD low VA- (Normal Operation) VD+	IA+ IA- ID+	- - -	25 -25 30	35 -35 45	- - -	25 -25 30	35 -35 45	- - -	25 -25 30	35 -35 50	mA mA mA
Power Supply Current (VA+)+(VL+) with APD, DPD high VA- (Power-Down Mode) VD+	IA+ IA- ID+	- - -	10 -10 10	50 -50 400	- - -	10 -10 10	50 -50 400	- - -	10 -10 10	50 -50 400	µA µA µA
Power Consumption (APD, DPD Low) (APD, DPD High)	PDN PDS	- -	400 0.15	575 2.5	- -	400 0.15	575 2.5	- -	400 0.15	600 2.5	mW mW
Power Supply (dc to 26 kHz) Rejection Ratio (26 kHz to 3.046 MHz)	PSRR	- -	54 100	- -	- -	54 100	- -	- -	54 100	- -	dB dB

Notes:
1. This parameter is guaranteed by design and/or characterization.
2. After calibration with DCAL connected to ACAL, and ZEROL & ZEROR terminated to AGND with an impedance matched to the AINR & AINL source impedance. Executing a calibration with ACAL tied low (See Power Down and Offset Calibration section) will yield an offset error of typically less than ± 5LSB.

DIGITAL FILTER CHARACTERISTICS

(T_A = 25 ° C; VA+, VL+ ,VD+ = 5V ± 5%; VA- = -5V ± 5%; Output word rate of 48 kHz)

Parameter			Symbol	Min	Typ	Max	Units
Passband	(-3 dB)	CS5336		0	to	22	kHz
	(-3 dB)	CS5338, CS5339		0	to	24	kHz
	(-0.01 dB)	CS5336		0	to	20	kHz
	(-0.01 dB)	CS5338, CS5339		0	to	22	kHz
Passband Ripple				-	-	± 0.01	dB
Stopband		CS5336		26	to	3046	kHz
		CS5338, CS5339		28	to	3044	kHz
Stopband Attenuation		(Note 3)		80	-	-	dB
Group Delay (OWR = Output Word Rate)			t_{gd}	-	18/OWR	-	s
Group Delay Variation vs. Frequency			Δt_{gd}	-	-	0.0	us

Notes: 3. The analog modulator samples the input at 3.072MHz for an output word rate of 48 kHz. There is no rejection of input signals which are multiples of the sampling frequency (that is: there is no rejection for n x 3.072MHz ±22kHz for the CS5338 & CS5339, or n x 3.072MHz ±20.0kHz for the CS5336, where n = 0,1,2,3...).

DIGITAL CHARACTERISTICS

(T_A = 25 °C; VA+, VL+ ,VD+ = 5V ± 5%; VA- = -5V ± 5%)

Parameter	Symbol	Min	Typ	Max	Units
High-Level Input Voltage	V_{IH}	70%VD+	-	-	V
Low-Level Input Voltage	V_{IL}	-	-	30% VD+	V
High-Level Output Voltage at Io = -20uA	V_{OH}	4.4	-	-	V
Low-Level Output Voltage at Io = 20uA	V_{OL}	-	-	0.1	V
Input Leakage Current	Iin	-	1.0	-	uA

ABSOLUTE MAXIMUM RATINGS (AGND, LGND, DGND = 0V, all voltages with respect to GND)

Parameter		Symbol	Min	Max	Units
DC Power Supplies:	Positive Analog	VA+	-0.3	+6.0	V
	Negative Analog	VA-	+0.3	-6.0	V
	Positive Logic	VL+	-0.3	(VA+) + 0.3	V
	Positive Digital	VD+	-0.3	+6.0	V
Input Current, Any Pin Except Supplies		I_{in}	-	± 10	mA
Analog Input Voltage (AIN and ZERO pins)		V_{INA}	(VA-)- 0.3	(VA+)+ 0.3	V
Digital Input Voltage		V_{IND}	-0.3	(VD+) + 0.3	V
Ambient Temperature (power applied)		T_A	-55	+125	°C
Storage Temperature		T_{stg}	-65	+150	°C

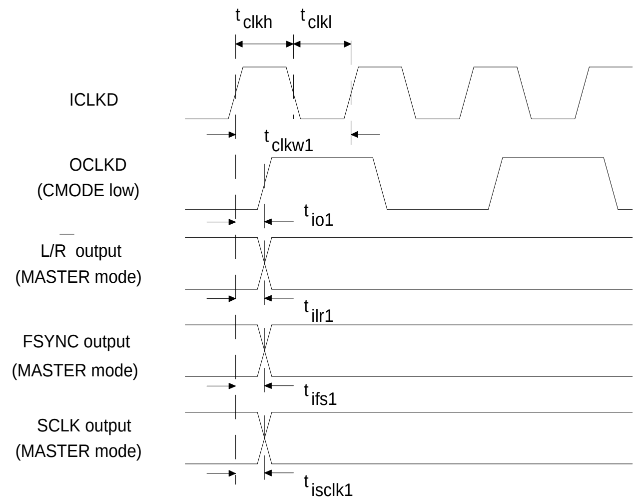
WARNING: Operation at or beyond these limits may result in permanent damage to the device. Normal operation is not guaranteed at these extremes.

SWITCHING CHARACTERISTICS

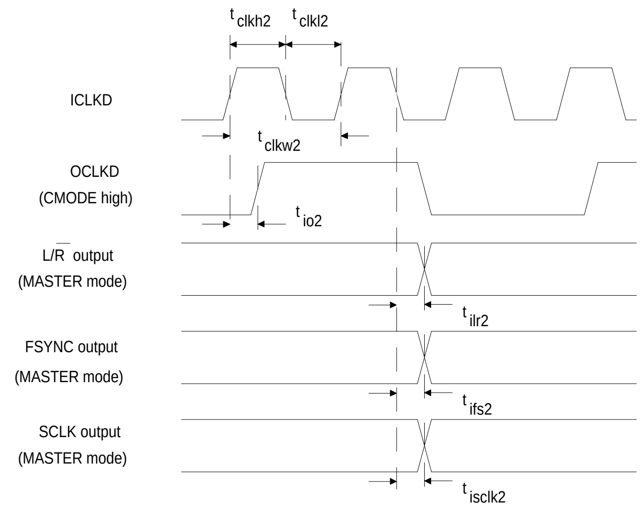
(T_A = 25 °C; VA+, VL+, VD+ = 5V ± 5%; VA- = -5V ± 5%; Inputs: Logic 0 = 0V, Logic 1 = VD+; C_L = 20 pF)

Parameter	Symbol	Min	Typ	Max	Unit
ICLKD Period (CMODE low) (Note 6)	t_{clkw1}	78	-	3906	ns
ICLKD Low (CMODE low)	t_{clkl1}	31	-	-	ns
ICLKD High (CMODE low)	t_{clkh1}	31	-	-	ns
ICLKD rising to OCLKD rising (CMODE low)	t_{io1}	5	-	40	ns
ICLKD Period (CMODE high)	t_{clkw2}	52	-	2604	ns
ICLKD Low (CMODE high)	t_{clkl2}	20	-	-	ns
ICLKD High (CMODE high)	t_{clkh2}	20	-	-	ns
ICLKD rising or falling to OCLKD rising (CMODE high, Note 4)	t_{io2}	5	-	45	ns
ICLKD rising to L/$\overline{R}$ edge (CMODE low, MASTER mode)	t_{ilr1}	5	-	50	ns
ICLKD rising to FSYNC edge (CMODE low, MASTER mode)	t_{ifs1}	5	-	50	ns
ICLKD rising to SCLK edge (CMODE low, MASTER mode)	t_{isclk1}	5	-	50	ns
ICLKD falling to L/$\overline{R}$ edge (CMODE high, MASTER mode)	t_{ilr2}	5	-	50	ns
ICLKD falling to FSYNC edge (CMODE high, MASTER mode)	t_{ifs2}	5	-	50	ns
ICLKD falling to SCLK edge (CMODE high, MASTER mode)	t_{isclk2}	5	-	50	ns
SCLK rising to SDATA valid (MASTER mode, Note 5)	t_{sdo}	0	-	50	ns
SCLK duty cycle (MASTER mode)		40	50	60	%
SCLK rising to L/$\overline{R}$ (MASTER mode, Note 5)	t_{mslr}	-20	-	20	ns
SCLK rising to FSYNC (MASTER mode, Note 5)	t_{msfs}	-20	-	20	ns
SCLK Period (SLAVE mode)	t_{sclkw}	155	-	-	ns
SCLK Pulse Width Low (SLAVE mode)	t_{sclkl}	60	-	-	ns
SCLK Pulse Width High (SLAVE mode)	t_{sclkh}	60	-	-	ns
SCLK rising to SDATA valid (SLAVE mode, Note 5)	t_{dss}	-	-	50	ns
L/$\overline{R}$ edge to MSB valid (SLAVE mode)	t_{lrdss}	-	-	50	ns
Falling SCLK to L/$\overline{R}$ edge delay (SLAVE mode, Note 5)	t_{slr1}	30	-	-	ns
L/$\overline{R}$ edge to falling SCLK setup time (SLAVE mode, Note 5)	t_{slr2}	30	-	-	ns
Falling SCLK to rising FSYNC delay (SLAVE mode, Note 5)	t_{sfs1}	30	-	-	ns
Rising FSYNC to falling SCLK setup time (SLAVE mode, Note 5)	t_{sfs2}	30	-	-	ns
DPD pulse width	t_{pdw}	2 x tclkw	-	-	ns
DPD rising to DCAL rising	t_{pcr}	-	-	50	ns
DPD falling to DCAL falling (OWR = Output Word Rate)	t_{pcf}	-	4096	-	1/OWR

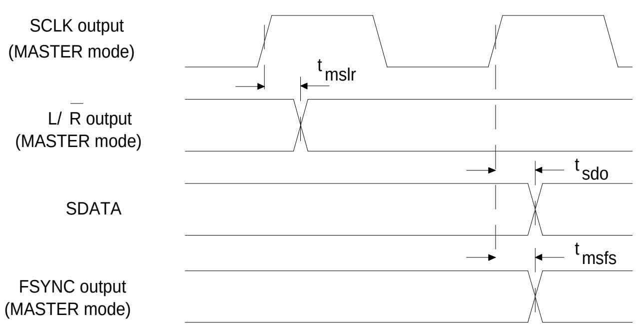
Notes: 4. ICLKD rising or falling depends on DPD to L/$\overline{R}$ timing (see Figure 2).
5. SCLK is shown for CS5336, CS5338. SCLK is inverted for CS5339.
6. Specifies minimum output word rate (OWR) of 1 kHz.



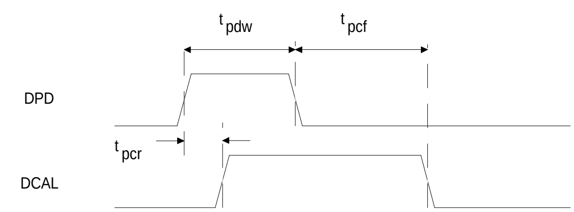
ICLKD to Outputs Propagation Delays (CMODE low)



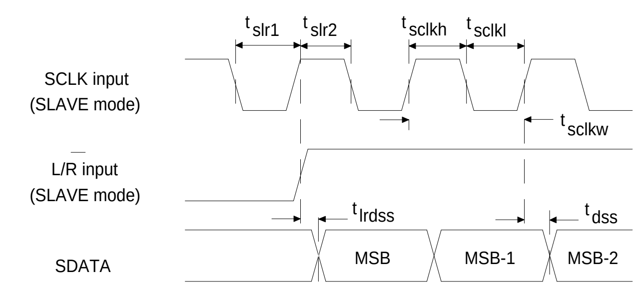
ICLKD to Outputs Propagation Delays (CMODE high)



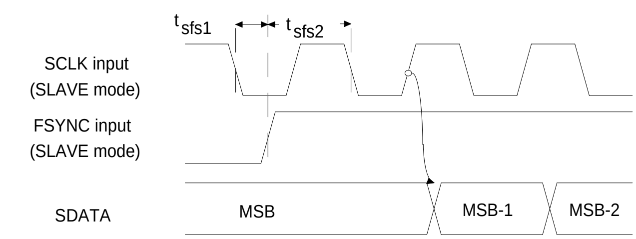
SCLK to SDATA, $L/\overline{R}$ & FSYNC - MASTER Mode



Power Down & Calibration Timing



SCLK to $L/\overline{R}$ & SDATA - SLAVE mode, FSYNC high



FSYNC to SCLK - SLAVE Mode, FSYNC Controlled.

RECOMMENDED OPERATING CONDITIONS

(AGND, LGND, DGND = 0V; all voltages with respect to ground)

Parameter		Symbol	Min	Typ	Max	Units
DC Power Supplies:	Positive Digital	VD+	4.75	5.0	VA+	V
	Positive Logic	VL+	4.75	5.0	VA+	V
	Positive Analog	VA+	4.75	5.0	5.25	V
	Negative Analog	VA-	−4.75	−5.0	−5.25	V
Analog Input Voltage	(Note 7)	V_{AIN}	−3.68	-	3.68	V

Notes: 7. The ADCs accept input voltages up to the analog supplies (VA+, VA-). They will produce a positive full-scale output for inputs above 3.68 V and negative full-scale output for inputs below -3.68 V. These values are subject to the gain error tolerance specification. Additional tag bits are output to indicate the amount of overdrive.

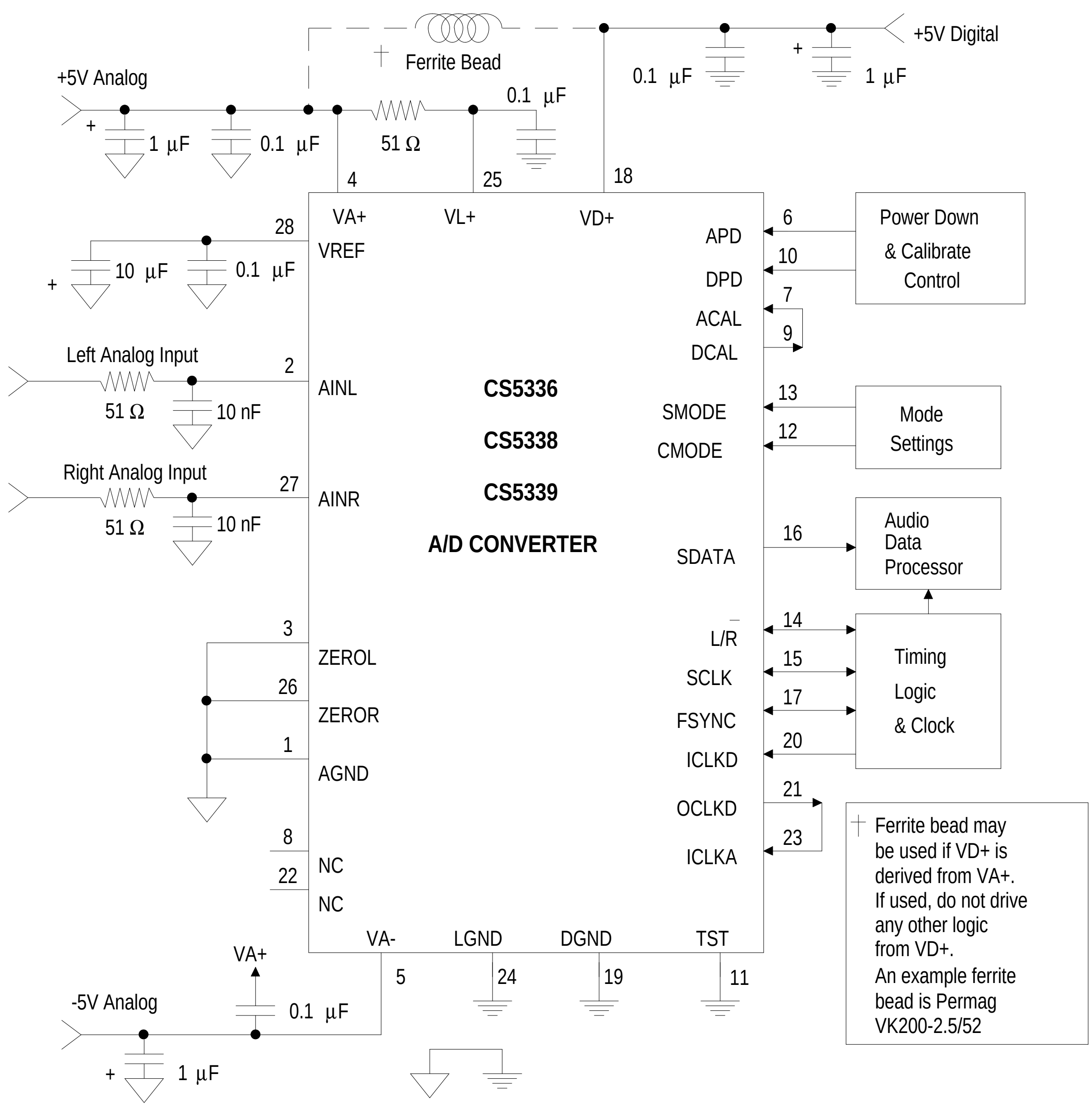


Figure 1. Typical Connection Diagram

GENERAL DESCRIPTION

The CS5336, CS5338, and CS5339 are 16-bit, 2-channel A/D converters designed specifically for stereo digital audio applications. The devices use two one-bit delta-sigma modulators which simultaneously sample the analog input signals at a 64 X sampling rate. The resulting serial bit streams are digitally filtered, yielding pairs of 16-bit values. This technique yields nearly ideal conversion performance independent of input frequency and amplitude. The converters do not require difficult-to-design or expensive anti-alias filters, and do not require external sample-and-hold amplifiers or a voltage reference.

An on-chip voltage reference provides for an input signal range of ± 3.68 volts. Any zero offset is internally calibrated out during a power-up self-calibration cycle. Output data is available in serial form, coded as 2's complement 16-bit numbers. Typical power consumption of only 400 mW can be further reduced by use of the power-down mode.

For more information on delta-sigma modulation and the particular implementation inside these ADCs, see the references at the end of this data sheet.

L/$\overline{R}$ (kHz)	CMODE	ICLKD (MHz)	OCLKD/ ICLKA (MHz)	SCLK (MHz)
32	low	8.192	4.096	2.048
32	high	12.288	4.096	2.048
44.1	low	11.2896	5.6448	2.8224
44.1	high	16.9344	5.6448	2.8224
48	low	12.288	6.144	3.072
48	high	18.432	6.144	3.072

Table 1. Common Clock Frequencies

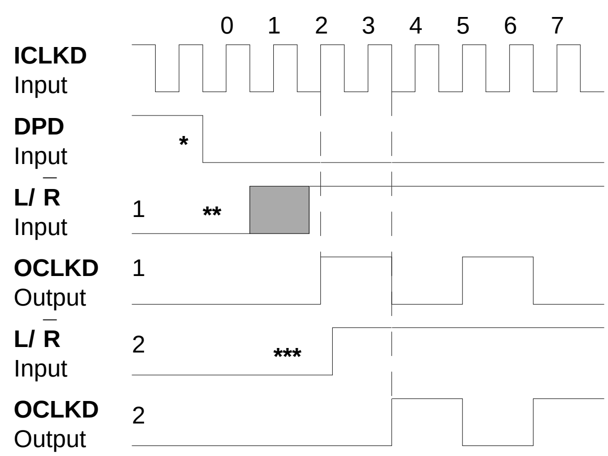
SYSTEM DESIGN

Very few external components are required to support the ADC. Normal power supply decoupling components, voltage reference bypass capacitors and a single resistor and capacitor on each input for anti-aliasing are all that's required, as shown in Figure 1.

Master Clock Input

The master input clock (ICLKD) into the ADC runs the digital filter, and is used to generate the modulator sampling clock. ICLKD frequency is determined by the desired Output Word Rate (OWR) and the setting of the CMODE pin. CMODE high will set the required ICLKD frequency to 384 X OWR, while CMODE low will set the required ICLKD frequency to 256 X OWR. Table 1 shows some common clock frequencies. The digital output clock (OCLKD) is always equal to 128 X OWR, which is always 2 X the input sample rate. OCLKD should be connected to ICLKA, which controls the input sample rate.

The phase alignment between ICLKD and OCLKD is determined as follows: when CMODE is



* D$\overline{P}$D low is recognized on the next ICLKD rising edge (#0)
** L/$\overline{R}$ rising before ICLKD rising #2 causes OCLKD -1
*** L/$\overline{R}$ rising after ICLKD rising #2 causes OCLKD - 2

Figure 2. ICLKD to OCLKD Timing with CMODE high (384 X OWR)

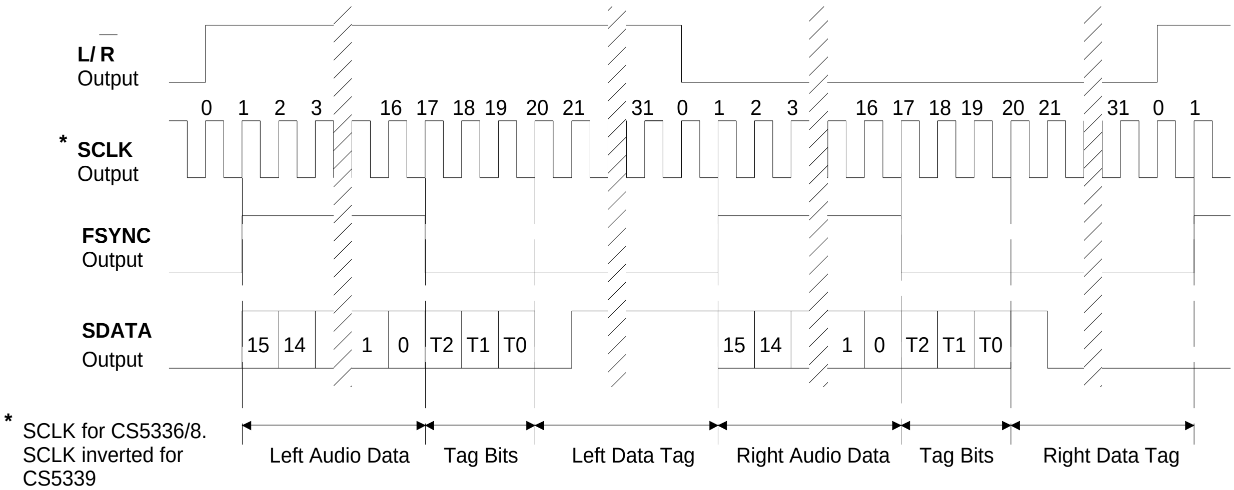


Figure 3. Data Output Timing - MASTER mode

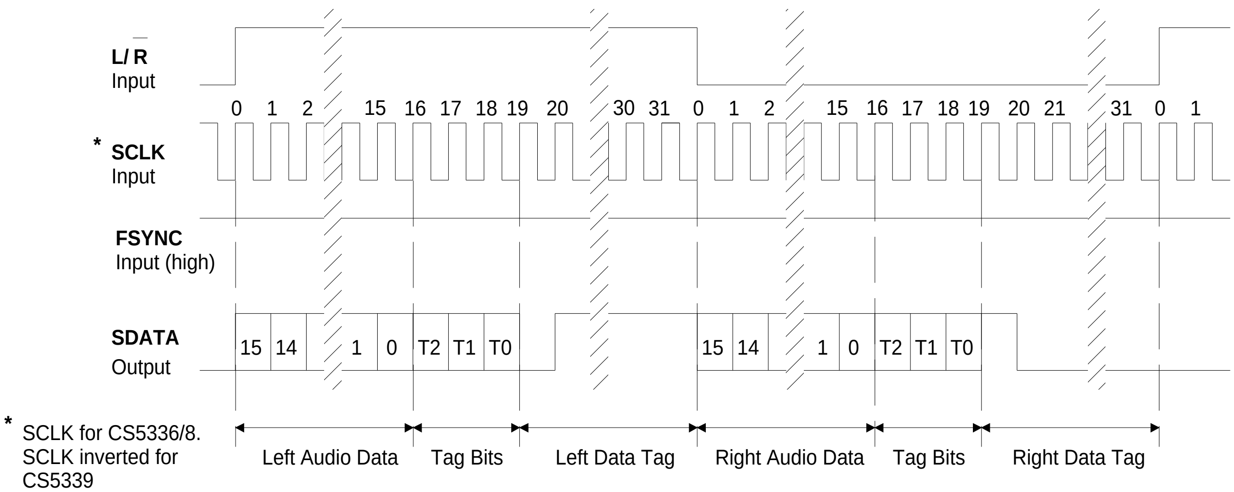


Figure 4. Data Output Timing - SLAVE Mode, FSYNC high

low, ICLKD is divided by 2 to generate OCLKD. The phase relationship between ICLKD and OCLKD is always the same, and is shown in the Switching Characteristics Timing Diagrams. When CMODE is high, OCLKD is ICLKD divided by 3. There are two possible phase relationships between ICLKD and OCLKD, which depend on the start-up timing between DPD and L/$\overline{R}$, shown in Figure 2.

Serial Data Interface

The serial data output interface has 3 possible modes of operation: MASTER mode, SLAVE mode with FSYNC high, and SLAVE mode with FSYNC controlled. In MASTER mode, the A/D converter is driven from a master clock (ICLKD) and outputs all other clocks, derived from ICLKD (see Figure 3). Notice the one SCLK cycle delay between L/$\overline{R}$ edges and FSYNC rising edges. FSYNC brackets the 16 data bits for each channel.

In SLAVE mode, L/$\overline{R}$ and SCLK are inputs. L/$\overline{R}$ must be externally derived from ICLKD, and should be equal to the Output Word Rate. SCLK should be equal to the input sample rate, which is equal to OCLKD/2. Other SCLK frequencies are possible, but may degrade dynamic range because of interference effects. Data bits are clocked out via the SDATA pin using the SCLK and L/$\overline{R}$ inputs. The rising edge of SCLK causes the ADC to

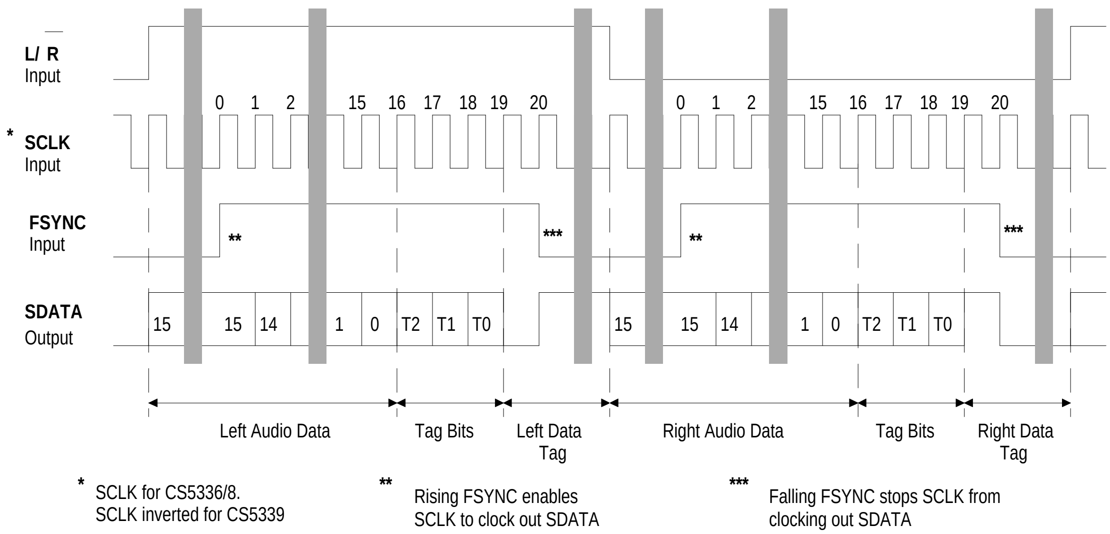


Figure 5. Data Output Timing - SLAVE Mode, FSYNC controlled

output each bit, except the MSB, which is clocked out by the L/$\overline{R}$ edge. As shown in Figure 4, when FSYNC is high, serial data bits are clocked immediately following the L/$\overline{R}$ edge.

In SLAVE mode with FSYNC controlled, as shown in Figure 5, when FSYNC is low, only the MSB is clocked out after the L/$\overline{R}$ edge. With FSYNC low, SCLK is ignored. When it is desired to start clocking out data, bring FSYNC high which enables SCLK to start clocking out data. Bringing FSYNC low will stop the data being clocked out. This feature is particularly useful to position in time the data bits onto a common serial bus.

Input Level	T2	T1	T0
1.375 x FS	1	1	1
1.250 x FS to 1.375 x FS	1	1	0
1.125 x FS to 1.250 x FS	1	0	1
1.000 x FS to 1.125 x FS	1	0	0
-1.006dB to 0.000dB	0	1	1
-3.060dB to -1.006dB	0	1	0
-6.000dB to -3.060dB	0	0	1
< -6.000dB	0	0	0

FS = Full Scale (0dB) Input

Table 2. Tag Bit Definition

The serial nature of the output data results in the left and right data words being read at different times. However, the words within an L/$\overline{R}$ cycle represent simultaneously sampled analog inputs.

In all modes, additional bits are output after the data bits: 3 tag bits and a left/right indicator. The tag bits indicate a near-to-clipping input condition for the data word to which the tag bits are attached. Table 2 shows the relationship between input level and the tag bit values. The serial bit immediately following the tag bits is 0 for the left channel, and 1 for the right channel. The remaining bits before the next L/$\overline{R}$ edge will be 1's for the left channel and 0's for the right channel. Normally, the tag bits are separated from the audio data by the digital signal processor. However, if the tag bits are interpreted as audio data, their position below the LSB would result as a very small dc offset.

In all modes, SCLK is shown for the CS5336 and CS5338, where data bits are clocked out on rising edges. SCLK is inverted for the CS5339.

Certain serial modes align well with various interface requirements. A CS5339 in MASTER mode, with an inverted L/$\overline{R}$ signal, generates I^2S (Philips) compatible timing. A CS5336 in MASTER mode, using FSYNC, interfaces well with a Motorola DSP56000. A CS5336 in SLAVE mode emulates a CS5326 style interface, and also links up to a DSP56000 in network mode.

Analog Connections

The analog inputs are presented to the modulators via the AINR and AINL pins. The analog input signal range is determined by the internal voltage reference value, which is typically -3.68 volts. The input signal range therefore is typically ± 3.68 volts.

The ADC samples the analog inputs at 3.072 MHz for a 12.288 MHz ICLKD (CMODE low). For the CS5336, the digital filter rejects all noise between 26 kHz and (3.072 MHz-26 kHz). For the CS5338 and CS5339, the digital filter rejects all noise between 28 kHz and (3.072 MHz-28 kHz). However, the filter will not reject frequencies right around 3.072 MHz (and multiples of 3.072 MHz). Most audio signals do not have significant energy at 3.072 MHz. Nevertheless, a 51 Ω resistor in series with the analog input, and a 10 nF NPO or COG capacitor to ground will attenuate any noise energy at 3.072 MHz, in addition to providing the optimum source impedance for the modulators. The use of capacitors which have a large voltage coefficient (such as general purpose ceramics) should be avoided since these can degrade signal linearity. If active circuitry precedes the ADC, it is recommended that the above RC filter is placed between the active circuitry and the AINR and AINL pins. The above example frequencies scale linearly with output word rate.

The on-chip voltage reference output is brought out to the VREF pin. A 10 µF electrolytic capacitor in parallel with a 0.1 µF ceramic capacitor attached to this pin eliminates the effects of high frequency noise. Note the negative value of VREF when using polarized capacitors. No load current may be taken from the VREF output pin.

The analog input level used as zero during the offset calibration period (described later) is input on the ZEROL and ZEROR pins. Typically, these pins are directly attached to AGND. For the ultimate in offset nulling, networks can be attached to ZEROR and ZEROL whose impedances match the impedances present on AINL and AINR.

Power-Down and Offset Calibration

The ADC has a power-down mode wherein typical consumption drops to 150 µW. In addition, exiting the power-down state initiates an offset calibration procedure.

APD and DPD are the analog and digital power-down pins. When high, they place the analog and digital sections in the power-down mode. Bringing these pins low takes the part out of power-down mode. DPD going low initiates a calibration cycle. If not using the power down feature, APD should be tied to AGND. When using the power down feature, DPD and APD may be tied together if the capacitor on VREF is not

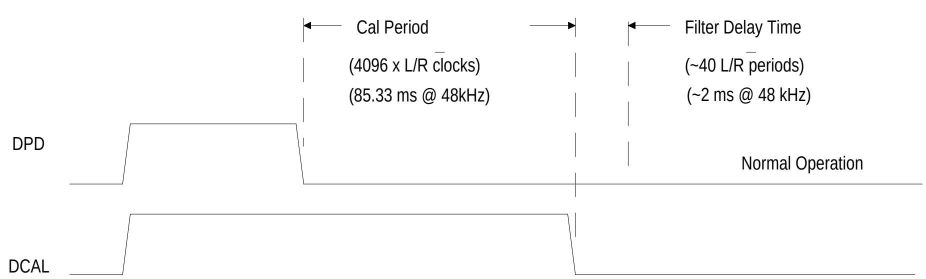


Figure 6. Initial Calibration Cycle Timing

greater than 10 µF, as stated in the "Power-Up Considerations" section.

During the offset calibration cycle, the digital section of the part measures and stores the value of the calibration input of each channel in registers. The calibration input value is subtracted from all future outputs. The calibration input may be obtained from either the analog input pins (AINL and AINR) or the zero pins (ZEROL and ZEROR) depending on the state of the ACAL pin. With ACAL low, the analog input pin voltages are measured, and with ACAL high, the zero pin voltages are measured.

As shown in Figure 6, the DCAL output is high during calibration, which takes 4096 L/$\overline{R}$ clock cycles. If DCAL is connected to the ACAL input, the calibration routine will measure the voltage on ZEROR and ZEROL. These should be connected directly to ground or through a network matched to that present on the analog input pins. Internal offsets of each channel will thus be measured and subsequently subtracted.

Alternatively, ACAL may be permanently connected low and DCAL utilized to control a multiplexer which grounds the user's front end. In this case, the calibration routine will measure and store not only the internal offsets but also any offsets present in the front end input circuitry.

During calibration, the digital output of both channels is forced to a 2's complement zero. Subtraction of the calibration input from conversions after calibration substantially reduces any power-on click that might otherwise be experienced. A short delay of approximately 40 output words will occur following calibration for the digital filter to begin accurately tracking audio band signals.

Power-up Considerations

Upon initial application of power to the supply pins, the data in the calibration registers will be indeterminate. A calibration cycle should always be initiated after application of power to replace potentially large values of data in these registers with the correct values.

The modulators settle very quickly (a matter of microseconds) after the analog section is powered on, either through the application of power, or by exiting the power-down mode. The voltage reference can take a much longer time to reach a final value due to the presence of large external capacitance on the VREF pin; allow approximately 5 ms/µF. The calibration period is long enough to allow the reference to settle for capacitor values of up to 10 µF. If a larger capacitor is used, additional time between APD going low and DPD going low should be allowed for VREF settling before a calibration cycle is initiated.

Grounding and Power Supply Decoupling

As with any high resolution converter, the ADC requires careful attention to power supply and grounding arrangements if its potential performance is to be realized. Figure 1 shows the recommended power arrangements, with VA+, VA- and VL+ connected to a clean ± 5 V supply. VD+, which powers the digital filter, may be run from the system +5V logic supply, provided that it is not excessively noisy (< ± 50 mV pk-to-pk). Alternatively, VD+ may be powered from VA+ via a ferrite bead. In this case, no additional devices should be powered from VD+. Analog ground and digital ground should be connected together near to where the supplies are brought onto the printed circuit board. Decoupling capacitors should be as near to the ADC as possible, with the low value ceramic capacitor being the nearest.

The printed circuit board layout should have separate analog and digital regions and ground planes, with the ADC straddling the boundary. All signals, especially clocks, should be kept away from the VREF pin in order to avoid unwanted coupling into the modulators. The VREF decoupling capacitors, particularly the 0.1 µF, must be positioned to minimize the electrical path from VREF to Pin 1 AGND and to minimize the path between VREF and the capacitors. An evaluation board is available which demonstrates the optimum layout and power supply arrangements, as well as allowing fast evaluation of the ADC.

To minimize digital noise, connect the ADC digital outputs only to CMOS inputs.

Synchronization of Multiple CS5336/8/9

In systems where multiple ADC's are required, care must be taken to insure that the ADC internal clocks are synchronized between converters to insure simultaneous sampling. In the absence of this synchronization, the sampling difference could be one ICLKD period which is typically 81.4 nsec for a 48 kHz sample rate.

SLAVE MODE

Synchronous sampling in the slave mode is achieved by connecting all DPD and APD pins to a single control signal and supplying the same ICLKD and $L/\overline{R}$ to all converters.

MASTER MODE

The internal counters of the CS5336/8/9 are reset during DPD/APD high and will start simultaneously by insuring that the release of DPD/APD for all converters is internally latched on the same rising edge of ICLKD. This can be achieved by connecting all DPD/APD pins to the same control signal and insuring that the DPD/APD falling edge occurs outside a ±30 ns window either side of an ICLKD rising edge.

PERFORMANCE

FFT Tests

For FFT based tests, a very pure sine wave is presented to the ADC, and an FFT analysis is performed on the output data. The resulting spectrum is a measure of the performance of the ADC.

Figure 7 shows the spectral purity of the CS5336 with a 1 kHz, -10 dB input. Notice the low noise floor, the absence of any harmonic distortion, and the Dynamic Range value of 95.41 dB.

Figure 8 shows the CS5336 high frequency performance. The input signal is a -10 dB, 9 kHz sine wave. Notice the small 2nd harmonic at 110 dB down.

Figure 9 shows the low-level performance of the CS5336. Notice the lack of any distortion components. Traditional R-2R ladder based ADC's can have problems with this test, since differential non-linearities around the zero point become very significant. Figure 10 shows the same very low input amplitude performance, but at 9kHz input frequency.

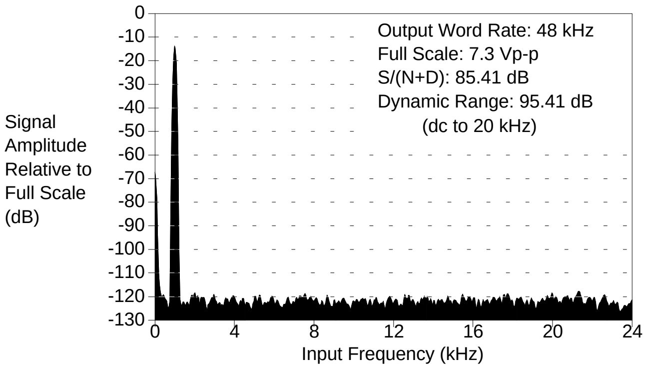


Figure 7. CS5336 FFT Plot with -10 dB, 1 kHz Input

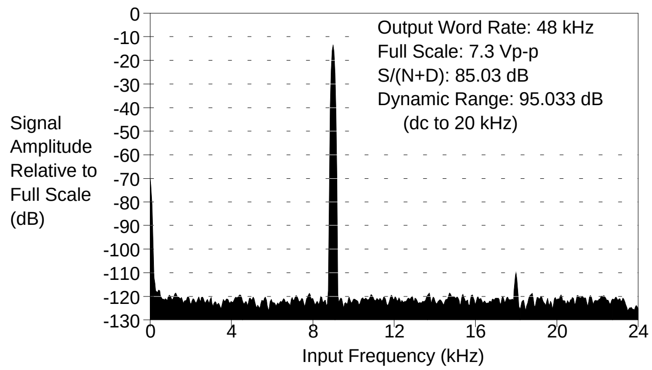


Figure 8. CS5336 FFT Plot with -10 dB, 9 kHz Input

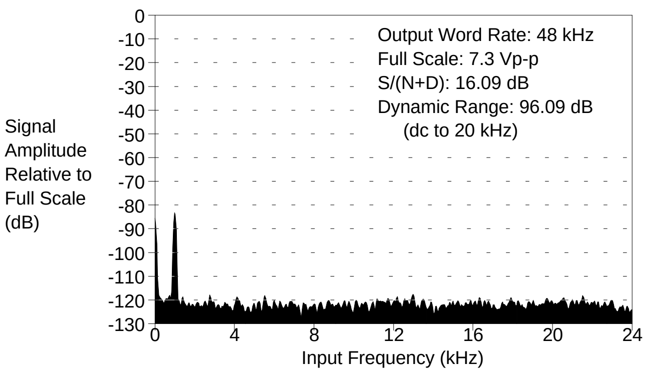


Figure 9. CS5336 FFT Plot with -80 dB, 1 kHz Input

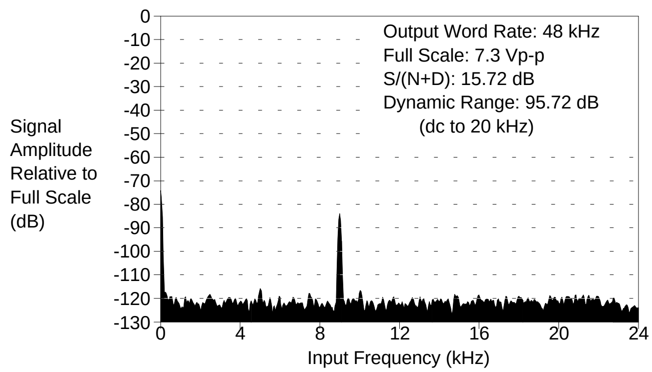


Figure 10. CS5336 FFT Plot with -80 dB, 9 kHz Input

DNL Tests

A Differential Non-Linearity test is also shown. Here, the converter is presented with a linear ramp signal. The resulting output codes are counted to yield a number which is proportional to the codewidth. A plot of codewidth versus code graphically illustrates the uniformity of the codewidths. Figure 11 shows the excellent Differential Non-Linearity of the CS5336. This plot displays the worst case positive and negative errors in each of 512 groups of 128 codes. Codewidths typically are within ± 0.2 LSB's of ideal. A delta-sigma modulator based ADC has no inherent mechanism for generating DNL errors. The residual small deviations shown in Figure 11 are a result of noise. Nevertheless, the performance shown is extremely good, and is superior to typical R-2R ladder based designs.

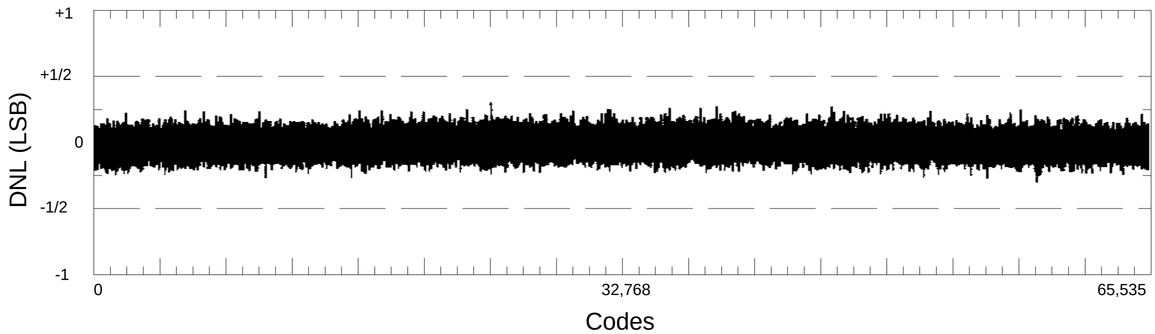


Figure 11. CS5336 Differential Non-Linearity Plot

Digital Filter

Figures 12 through 17 show the performance of the digital filter included in the ADC. All the plots assume an output word rate of 48 kHz. The filter frequency response will scale precisely with changes in output word rate. The passband ripple is flat to ± 0.01 dB maximum. Stopband rejection is greater than 80 dB.

Figures 12,14 &16 show the CS5338 and CS5339 filter characteristics. Figure 17 is an expanded view of the transition band.

Figures 13,15 & 17 show the CS5336 filter characteristics. Figure 17 is an expanded view of the transition band.

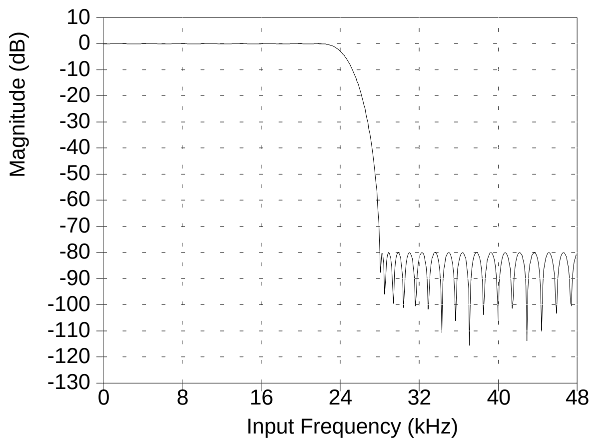


Figure 12. CS5338/9 Digital Filter Stopband Rejection

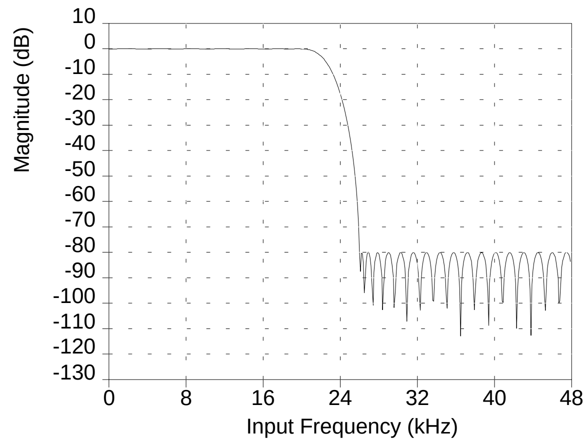


Figure 13. CS5336 Digital Filter Stopband Rejection

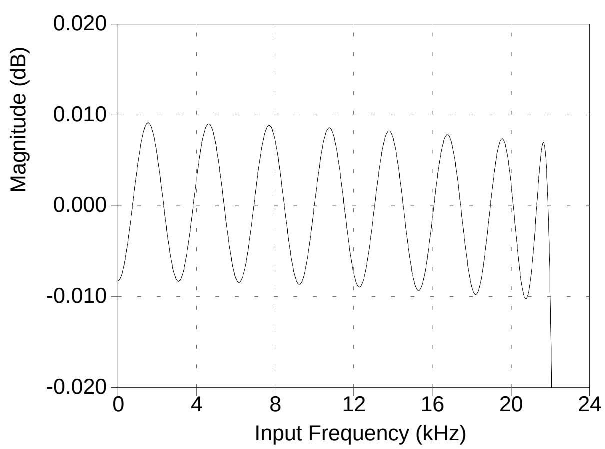


Figure 14. CS5338/9 Digital Filter Passband Ripple

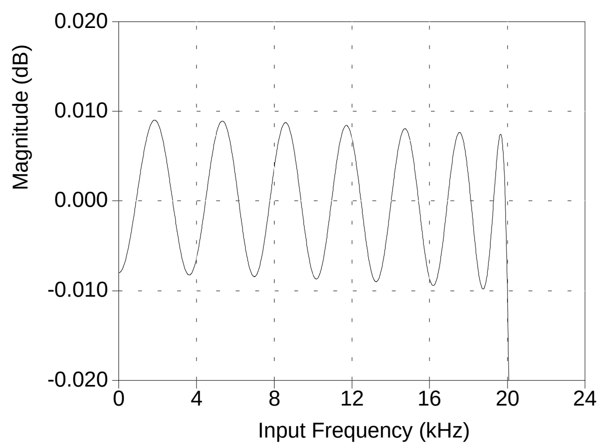


Figure 15. CS5336 Digital Filter Passband Ripple

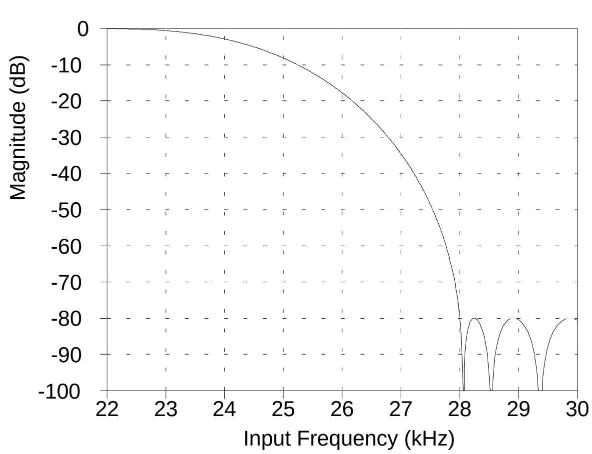


Figure 16. CS5338/9 Digital Filter Transition Band

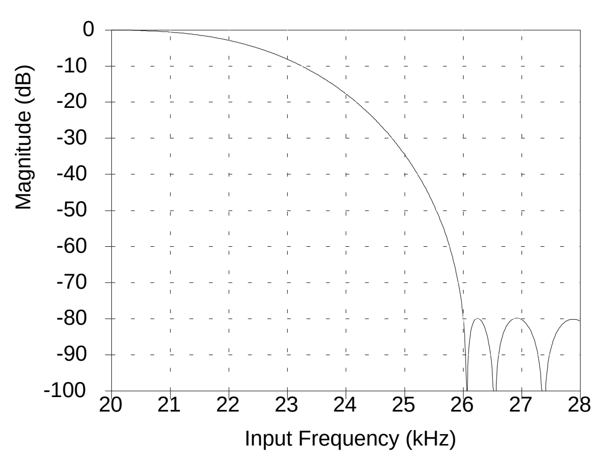


Figure 17. CS5336 Digital Filter Transition Band

PIN DESCRIPTIONS

ANALOG GROUND	**AGND**	1	28	**VREF**	VOLTAGE REFERENCE OUTPUT
LEFT CHANNEL ANALOG INPUT	**AINL**	2	27	**AINR**	RIGHT CHANNEL ANALOG INPUT
LEFT CHANNEL ZERO INPUT	**ZEROL**	3	26	**ZEROR**	RIGHT CHANNEL ZERO INPUT
POSITIVE ANALOG POWER	**VA+**	4	25	**VL+**	ANALOG SECTION LOGIC POWER
NEGATIVE ANALOG POWER	**VA-**	5	24	**LGND**	ANALOG SECTION LOGIC GROUND
ANALOG POWER DOWN INPUT	**APD**	6	23	**ICLKA**	ANALOG SECTION CLOCK INPUT
ANALOG CALIBRATE INPUT	**ACAL**	7	22	**NC**	NO CONNECT
NO CONNECT	**NC**	8	21	**OCLKD**	DIGITAL SECTION OUTPUT CLOCK
DIGITAL CALIBRATE OUTPUT	**DCAL**	9	20	**ICLKD**	DIGITAL SECTION CLOCK INPUT
DIGITAL POWER DOWN INPUT	**DPD**	10	19	**DGND**	DIGITAL GROUND
TEST	**TST**	11	18	**VD+**	DIGITAL SECTION POSITIVE POWER
SELECT CLOCK MODE	**CMODE**	12	17	**FSYNC**	FRAME SYNC SIGNAL
SELECT SERIAL I/O MODE	**SMODE**	13	16	**SDATA**	SERIAL DATA OUTPUT
LEFT/RIGHT SELECT	**$L/\overline{R}$**	14	15	**SCLK**	SERIAL DATA CLOCK

Power Supply Connections

VA+ - Positive Analog Power, PIN 4.
Positive analog supply. Nominally +5 volts.

VL+ - Positive Logic Power, PIN 25.
Positive logic supply for the analog section. Nominally +5 volts.

VA- - Negative Analog Power, PIN 5.
Negative analog supply. Nominally -5 volts.

AGND - Analog Ground, PIN 1.
Analog ground reference.

LGND - Logic Ground, PIN 24
Ground for the logic portions of the analog section.

VD+ - Positive Digital Power, PIN 18.
Positive supply for the digital section. Nominally +5 volts.

DGND - Digital Ground, PIN 19.
Digital ground for the digital section.

Analog Inputs

AINL, AINR - Left and Right Channel Analog Inputs, PINS 2, 27
Analog input connections for the left and right input channels. Nominally ±3.68 volts full scale.

ZEROL, ZEROR - Zero Level Inputs for Left and Right Channels, PINS 3, 26.

Analog zero level inputs for the left and right channels. The levels present on these pins can be used as zero during the offset calibration cycle. Normally connected to AGND, optionally through networks matched to the analog input networks.

Analog Outputs

VREF - Voltage Reference Output, PIN 28.

Nominally -3.68 volts. Normally connected to a 0.1µF ceramic capacitor in parallel with a 10µF or larger electrolytic capacitor. Note the negative output polarity.

Digital Inputs

ICLKA - Analog Section Input Clock, PIN 23.

This clock is internally divided by 2 to set the modulators' sample rate. Sampling rates, output rates, and digital filter characteristics scale to ICLKA frequency. ICLKA frequency is 128 X the output word rate. For example, 6.144 MHz ICLKA corresponds to an output word rate of 48 kHz per channel. Normally connected to OCLKD.

ICLKD - Digital Section Input Clock, PIN 20.

This is the clock which runs the digital filter. ICLKD frequency is determined by the required output word rate and by the CMODE pin. If CMODE is low, ICLKD frequency should be 256 X the desired output word rate. If CMODE is high, ICLKD should be 384 X the desired output word rate. For example, with CMODE low, ICLKD should be 12.288 MHz for an output word rate of 48 kHz. This clock also generates OCLKD, which is always 128 X the output word rate.

APD - Analog Power Down, PIN 6.

Analog section power-down command. When high, the analog circuitry is in power-down mode. APD is normally connected to DPD when using the power down feature. If power down is not used, then connect APD to AGND.

DPD - Digital Power Down, PIN 10

Digital section power-down command. Bringing DPD high puts the digital section into power-down mode. Upon returning low, the ADC starts an offset calibration cycle. This takes 4096 $L/\overline{R}$ periods (85.33 ms with a 12.288 MHz ICLKD). DCAL is high during the calibrate cycle and goes low upon completion. DPD is normally connected to APD when using the power down feature. A calibration cycle should always be initiated after applying power to the supply pins.

ACAL - Analog Calibrate, PIN 7.

Analog section calibration command. When high, causes the left and right channel modulator inputs to be internally connected to ZEROL and ZEROR inputs respectively. May be connected to DCAL.

CMODE - Clock Mode Select, PIN 12.

CMODE should be tied low to select an ICLKD frequency of 256 X the output word rate. CMODE should be tied high to select an ICLKD frequency of 384 X the output word rate.

SMODE - Serial Interface Mode Select, PIN 13.

SMODE should be tied high to select serial interface master mode, where SCLK, FSYNC and L/$\overline{\text{R}}$ are all outputs, generated by internal dividers operating from ICLKD. SMODE should be tied low to select serial interface slave mode, where SCLK, FSYNC and L/$\overline{\text{R}}$ are all inputs. In slave mode, L/$\overline{\text{R}}$, FSYNC and SCLK need to be derived from ICLKD using external dividers.

Digital Outputs

SDATA - Serial Data Output, PIN 16.

Audio data bits are presented MSB first, in 2's complement format. Additional tag bits, which indicate input overload and left/right channel data, are output immediately following each audio data word.

DCAL - Digital Calibrate Output, PIN 9.

DCAL rises immediately upon entering the power-down state (DPD brought high). It returns low 4096 L/$\overline{\text{R}}$ periods after leaving the power down state (DPD brought low), indicating the end of the offset calibration cycle (which = 85.33 ms with a 12.288 MHz ICLKD). May be connected to ACAL.

OCLKD - Digital Section Output Clock, PIN 21.

OCLKD is always 128 X the output word rate. Normally connected to ICLKA.

Digital Inputs or Outputs

SCLK - Serial Data Clock, PIN 15.

Data is clocked out on the rising edge of SCLK for the CS5336 and CS5338. Data is clocked out on the falling edge of SCLK for the CS5339.

In master mode (SMODE high), SCLK is a continuous output clock at 64 X the output word rate.

In slave mode (SMODE low), SCLK is an input, which requires a continuously supplied clock at any frequency from 32 X to 128 X the output word rate (64 X is preferred). When FSYNC is high, SCLK clocks out serial data, except for the MSB which appears on SDATA when L/$\overline{\text{R}}$ changes.

L/$\overline{R}$ - Left/Right Select, PIN 14.

In master mode (SMODE high), L/$\overline{R}$ is an output whose frequency is at the output word rate. L/$\overline{R}$ edges occur 1 SCLK cycle before FSYNC rises. When L/$\overline{R}$ is high, left channel data is on SDATA, except for the first SCLK cycle. When L/$\overline{R}$ is low, right channel data is on SDATA, except for the first SCLK cycle. The MSB data bit appears on SDATA one SCLK cycle after L/$\overline{R}$ changes.

In slave mode (SMODE low), L/$\overline{R}$ is an input which selects the left or right channel for output on SDATA. The rising edge of L/$\overline{R}$ starts the MSB of the left channel data. L/$\overline{R}$ frequency must be equal to the output word rate.

Although the outputs of each channel are transmitted at different times, the two words in an L/$\overline{R}$ cycle represent simultaneously sampled analog inputs.

FSYNC - Frame Synchronization Signal, PIN 17.

In master mode (SMODE high), FSYNC is an output which goes high coincident with the start of the first SDATA bit (MSB) and falls low immediately after the last SDATA audio data bit (LSB).

In slave mode (SMODE low), FSYNC is an input which controls the clocking out of the data bits on SDATA. FSYNC is normally tied high, which causes the data bits to be clocked out immediately following L/$\overline{R}$ transitions. If it is desired to delay the data bits from the L/$\overline{R}$ edge, then FSYNC must be low during the delay period. Bringing FSYNC high will then enable the clocking out of the SDATA bits. Note that the MSB will be clocked out based on the L/$\overline{R}$ edge, independent of the state of FSYNC.

Miscellaneous

NC - No Connection, PINS 8, 22.

These two pins are bonded out to test outputs. They must not be connected to any external component or any length of PC trace.

TST -Test Input, PIN 11.

Allows access to the ADC test modes, which are reserved for factory use. Must be tied to DGND.

PARAMETER DEFINITIONS

Resolution - The total number of possible output codes is equal to 2^N, where N = the number of bits in the output word for each channel.

Dynamic Range - Full scale (RMS) signal to broadband noise ratio. The broadband noise is measured over the specified bandwidth, and with an input signal 60dB below full-scale. Units in decibels.

Signal-to-(Noise plus Distortion) Ratio - The ratio of the rms value of the signal to the rms sum of all other spectral components over the specified bandwidth (typically 10 Hz to 20 kHz), including distortion components. Expressed in decibels.

Total Harmonic Distortion - The ratio of the rms sum of all harmonics up to 20 kHz to the rms value of the signal. Units in percent.

Interchannel Phase Deviation - The difference between the left and right channel sampling times.

Interchannel Isolation - A measure of crosstalk between the left and right channels. Measured for each channel at the converter's output with the input under test grounded and a full-scale signal applied to the other channel. Units in decibels.

Interchannel Gain Mismatch - The gain difference between left and right channels. Units in decibels.

Gain Error - The deviation of the measured full scale amplitude from the ideal full scale amplitude value.

Gain Drift - The change in gain value with temperature. Units in ppm/°C.

Bipolar Offset Error - The deviation of the mid-scale transition (111...111 to 000...000) from the ideal (1/2 LSB below AGND). Units in LSBs.

REFERENCES

1) "A Stereo 16-bit Delta-Sigma A/D Converter for Digital Audio" by D.R. Welland, B.P. Del Signore, E.J. Swanson, T. Tanaka, K. Hamashita, S. Hara, K. Takasuka. Paper presented at the 85th Convention of the Audio Engineering Society, November 1988.

2) " The Effects of Sampling Clock Jitter on Nyquist Sampling Analog-to-Digital Converters, and on Oversampling Delta Sigma ADC's" by Steven Harris. Paper presented at the 87th Convention of the Audio Engineering Society, October 1989.

3) " An 18-Bit Dual-Channel Oversampling Delta-Sigma A/D Converter, with 19-Bit Mono Application Example" by Clif Sanchez. Paper presented at the 87th Convention of the Audio Engineering Society, October 1989.

Ordering Guide

Model	Resolution	Passband	SCLK	Temperature	Package
CS5336-KP	16-bits	22 kHz	↑ active	0°C to 70 °C	28-pin Plastic DIP
CS5336-BP	16-bits	22 kHz	↑ active	-40 to +85 °C	28-pin Plastic DIP
CS5338-KP	16-bits	24 kHz	↑ active	0°C to 70 °C	28-pin Plastic DIP
CS5339-KP	16-bits	24 kHz	↓ active	0°C to 70 °C	28-pin Plastic DIP
CS5336-KS	16-bits	22 kHz	↑ active	0°C to 70 °C	28-pin SOIC
CS5336-BS	16-bits	22 kHz	↑ active	-40 to +85 °C	28-pin SOIC
CS5338-KS	16-bits	24 kHz	↑ active	0°C to 70 °C	28-pin SOIC
CS5339-KS	16-bits	24 kHz	↓ active	0°C to 70 °C	28-pin SOIC
CS5336-TC	16-bits	22 kHz	↑ active	-55 to +125 °C	28-pin Sidebrazed Ceramic DIP

CDB5336	CS5336 Evaluation Board
CDB5338	CS5338 Evaluation Board
CDB5339	CS5339 Evaluation Board

CRYSTAL
Semiconductor Corporation

CDB5336
CDB5338 CDB5339

Evaluation Board for CS5336, CS5338 & CS5339

Features

- Demonstrates recommended layout and grounding arrangements
- CS8402 Generates AES/EBU, S/PDIF & CP-340 Compatible Digital Audio
- Buffered Serial Output Interface
- 16-Bit Parallel Output Interface
- Digital and Analog Patch Areas
- On-board or externally supplied system timing

General Description

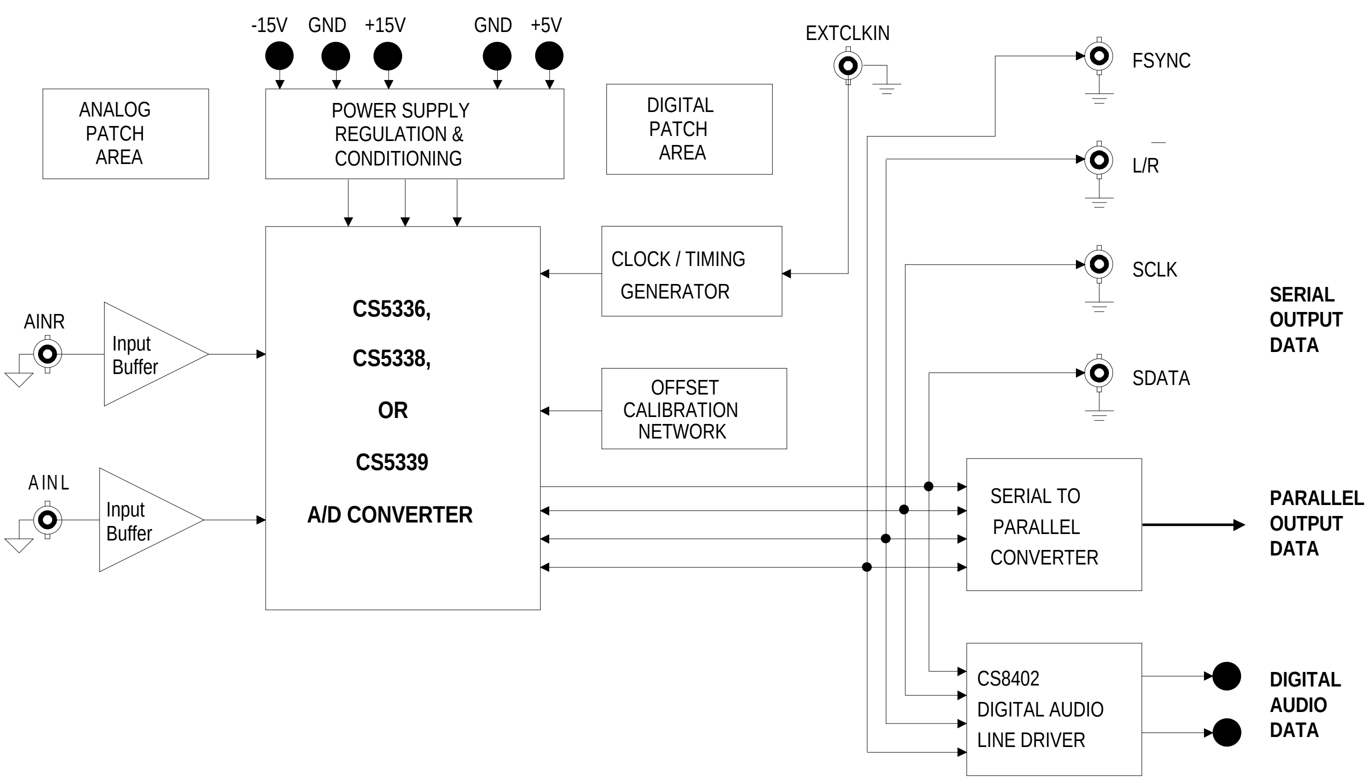
The CDB5336, CDB5338 & CDB5339 evaluation boards allow fast evaluation of the CS5336, CS5338 and CS5339 16-bit, stereo A/D converters. The boards generate all converter timing signals and provide both parallel and serial output interfaces. Evaluation requires a digital signal processor, a low-distortion signal source, and a power supply.

Also included is a CS8402 digital audio transmitter I.C., which can generate AES/EBU, S/PDIF & EIAJ CP-340 compatible audio data.

The evaluation boards may also be configured to accept external timing signals for operation in a user application during system development.

ORDERING INFORMATION:

CDB5336, CDB5338, CDB5339



Crystal Semiconductor Corporation
P.O. Box 17847, Austin, TX 78760
(512) 445 7222 FAX: (512) 445 7581

Power Supply Circuitry

The schematic diagram in Figure 1 shows the evaluation board power supply circuitry. Power is supplied to the evaluation board by five binding posts. The ±5 Volt analog power supply inputs of the converter are derived from ±15 Volts using the voltage regulators U10 and U11. The +5 Volt digital supply for the converter and the discrete logic on the board is provided by the +5V and DGND binding posts. D1, D2 and D4 are transient suppressors which also provide protection from incorrectly connected power supply leads. C25-C28, C30 and C31 provide general power supply filtering for the analog supplies. As shown in Figure 2, C10-C13 provide localized decoupling for the converter VA+ and VA- pins. Note that C13 is connected between VA- and VA+ and not VA- and AGND. Space for a ferrite bead inductor, L1, has been provided so that the board may be modified to power the converter's VD+ input directly from the VA+ supply. Note that the trace connecting the VD+ power to the VD+ of the converter must be broken before L1 may be installed. R5 and C7 low-pass filter the analog logic power supply pin, VL+. The evaluation board uses both an analog and a digital ground plane which are connected at a single point by J1. This ground plane arrangement isolates the board's digital logic from the analog circuitry.

Offset Calibration & Reset Circuit

Figure 1, shows the optional offset calibration circuit provided on the evaluation board. Upon power-up, this circuit provides a pulse on the Analog-to-Digital Converter's DPD pin initiating an offset calibration cycle. Releasing SW1 also initiates an offset calibration cycle. P6 (see Figure 2) selects the signal source used during offset calibration. In the "AIN" position, the AINL and AINR inputs are selected during calibration, while in the "ZERO" position, the ZEROL and ZEROR inputs are selected.

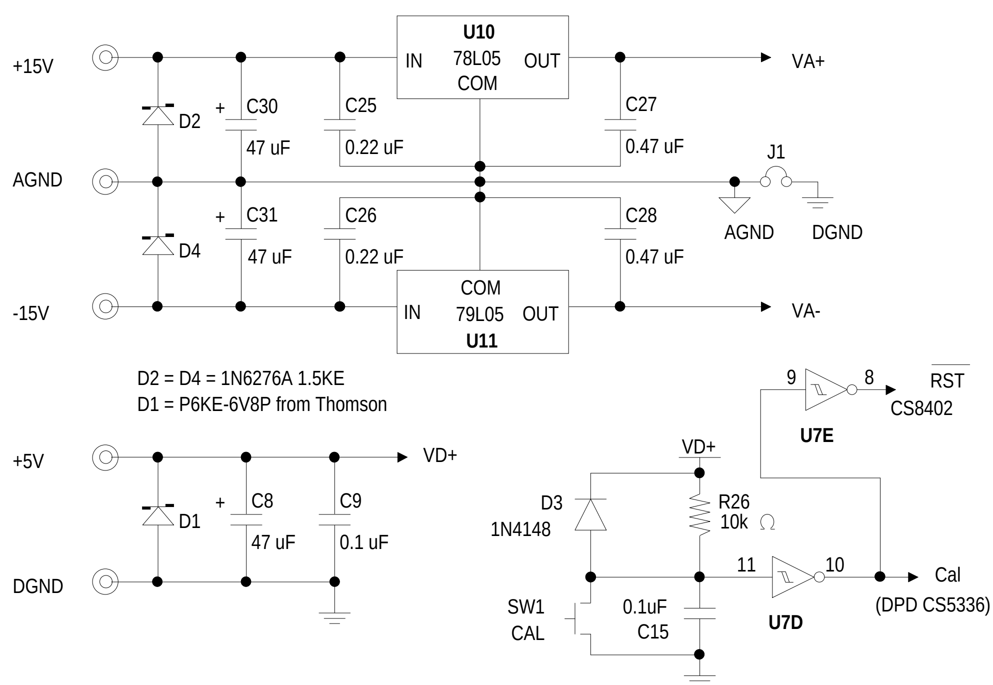


Figure 1. Power Supply and Reset Circuitry

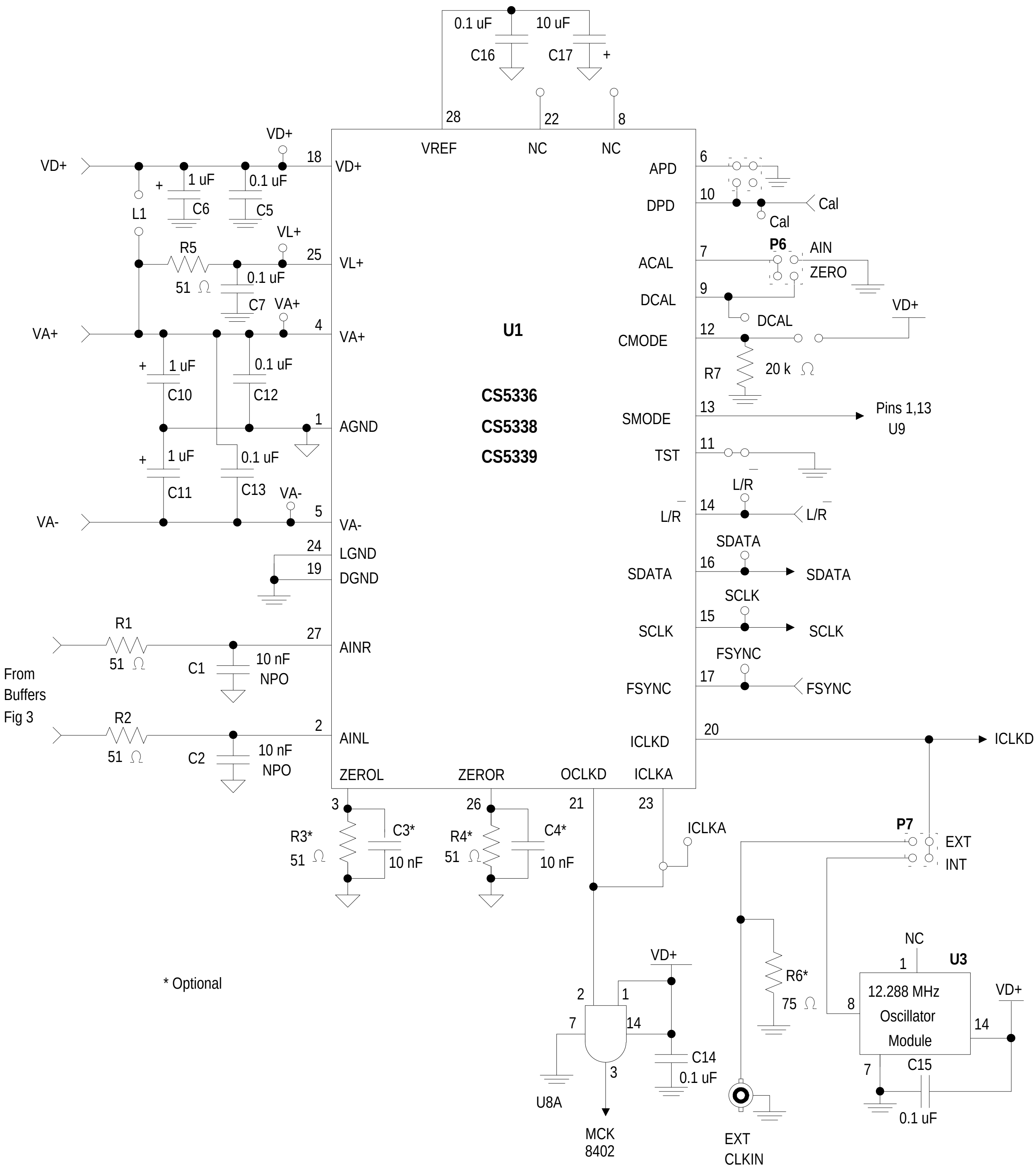


Figure 2 ADC Connections

Analog Inputs

As shown in Figure 2, the analog input signals are connected to the CS5336 via an RC network. R1 and C1 provide antialiasing and optimum source impedance for the right analog input channel while R2 and C2 do so for the left channel. The ZEROR and ZEROL inputs are tied to the analog ground plane on the board as shipped from the factory, but space is provided for an optional RC section on each. These RC sections may be added to model the output impedance of the analog signal source to minimize offset error during calibration.

Figure 3 shows the optional input buffer circuit. This can be used as an example input buffer circuit for your application. If the ADC is driven from a 50Ω source impedance signal generator, the input buffer amplifiers may be bypassed. Place P8 and P9 jumpers in the OUT position, and short circuit R1 and R2. This ensures that the ADC is driven from a 50Ω source resistance. Also remove U13 op-amp, to remove the 1kΩ load impedance.

Timing Generator

P7 selects the master clock source supplied to the ICLKD pin of the converter. As shipped from the factory, P7 is set to the "INT" position to select the 12.288 MHz clock signal provided by U3. An external master clock signal may be connected to the EXTCLKIN connector and selected by placing P7 in the "EXT" position. Note that R6, tied between EXTCLKIN and GND, is available for impedance matching an external clock source. The board is shipped with SMODE high, which selects MASTER timing mode. In this mode, SCLK, L/$\overline{\text{R}}$ and FSYNC are all outputs, generated by the converter from ICLKD.

Serial Output Interface

The serial output interface is provided by the SDATA, SCLK, FSYNC and L/$\overline{\text{R}}$ BNC connectors on the evaluation board. These out-

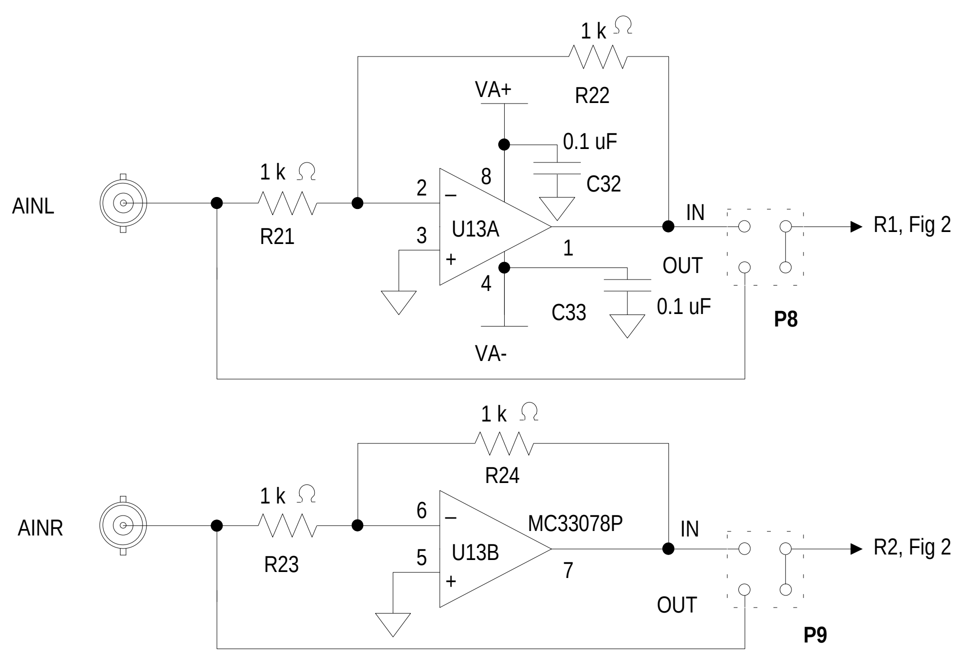


Figure 3. Input Buffer Circuit

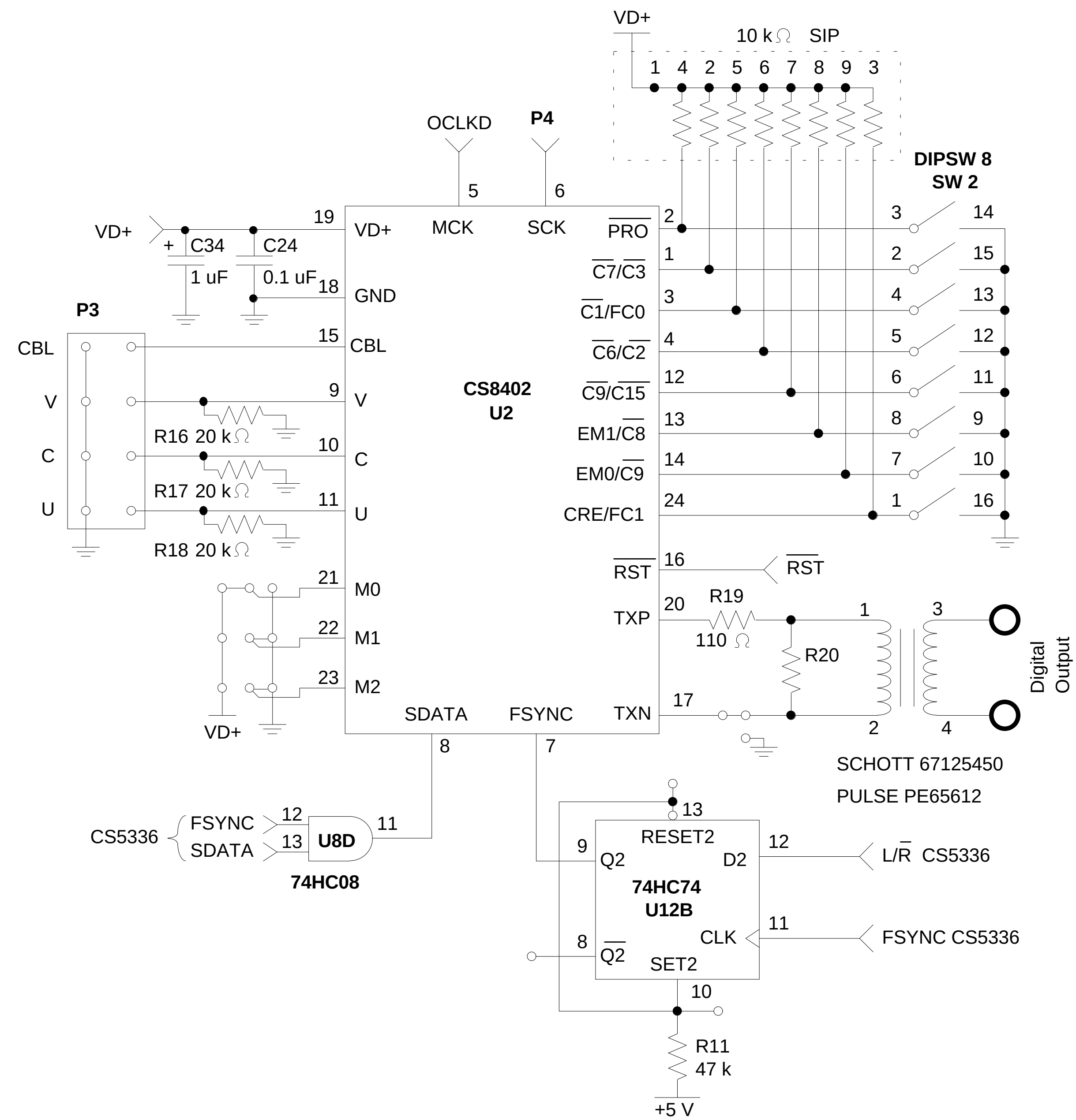


Figure 4. CS8402 Digital Audio Line Driver Connections

puts are buffered, as shown in Figure 5, in order to isolate the converter from the digital signal processor. If slave mode is selected by pulling SMODE low, then U9 (74HC243) will change to the opposite direction, and act as an input buffer. U9 is provided to protect against inadvertent external driving of SCLK, L/R̄ and FSYNC while in MASTER mode. U9 is not necessary in your application circuit.

Jumper P4 allows the board to be configured for either the CS5336/38, or the CS5339, which have opposite polarities of SCLK.

Parallel Output Interface

Figure 6 depicts the parallel output interface on the evaluation board. 16-bit words are assembled from the serial data output of the converter. Each bit of serial data is clocked out of the converter

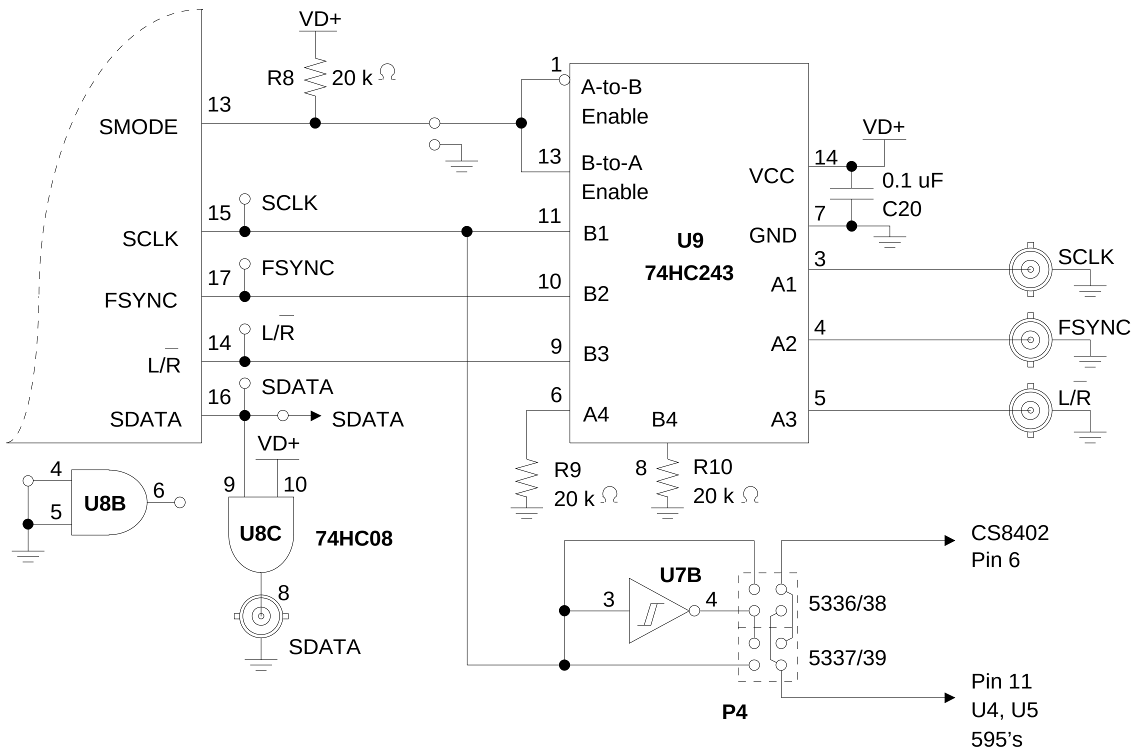


Figure 5. Serial Output Interface

on the rising edge of SCLK and shifted into the 16-bit shift register formed by U4 and U5 on SCLK's falling edge. After all data bits for the selected channel have been shifted into U4 and U5 the data is latched onto P1 by a delayed version of FSYNC.

P5 selects the channel whose output data will be converted to parallel form and presented on P1. With P5 in the "B" (both) position, parallel data from one channel will be presented first with data from the other channel presented subsequently. In the "L" (left) position, only left channel conversions will be presented, while in the "R" (right) position only right channel conversions are presented.

Two interface mechanisms are provided for reading the data from this port. With the first, the edges of L/$\overline{\mathrm{R}}$ may be used to clock the parallel data into the digital signal processor. (Set jumper P2 into the L/$\overline{\mathrm{R}}$ position.) Alternatively, a handshake protocol implemented with DACK and $\overline{\mathrm{DRDY}}$ may be used to transfer data to the signal processor. (Set jumper P2 to the $\overline{\mathrm{DRDY}}$ position.) The fall of $\overline{\mathrm{DRDY}}$ informs the digital signal processor that a new data word is available. The processor then reads the port and acknowledges the transfer by asserting DACK. Note that $\overline{\mathrm{DRDY}}$ will not be asserted again unless DACK is momentarily brought high although new data will continue to be latched onto the port.

Digital Audio Standard Interface

Included on the evaluation board is a CS8402 Digital Audio Line Driver. This device can implement AES/EBU, S/PDIF and EIAJ CP-340 interface standards. Figure 4 shows the schematic for the CS8402. P3 allows the C, U and V bits to be driven from external logic using the CBL output for block synchronization. SW2 provides 8 DIP switches to select various modes and bits for the CS8402. Table 3 lists the settings for the professional mode which is the default setting for the evaluation board from the factory. The third switch selects between professional

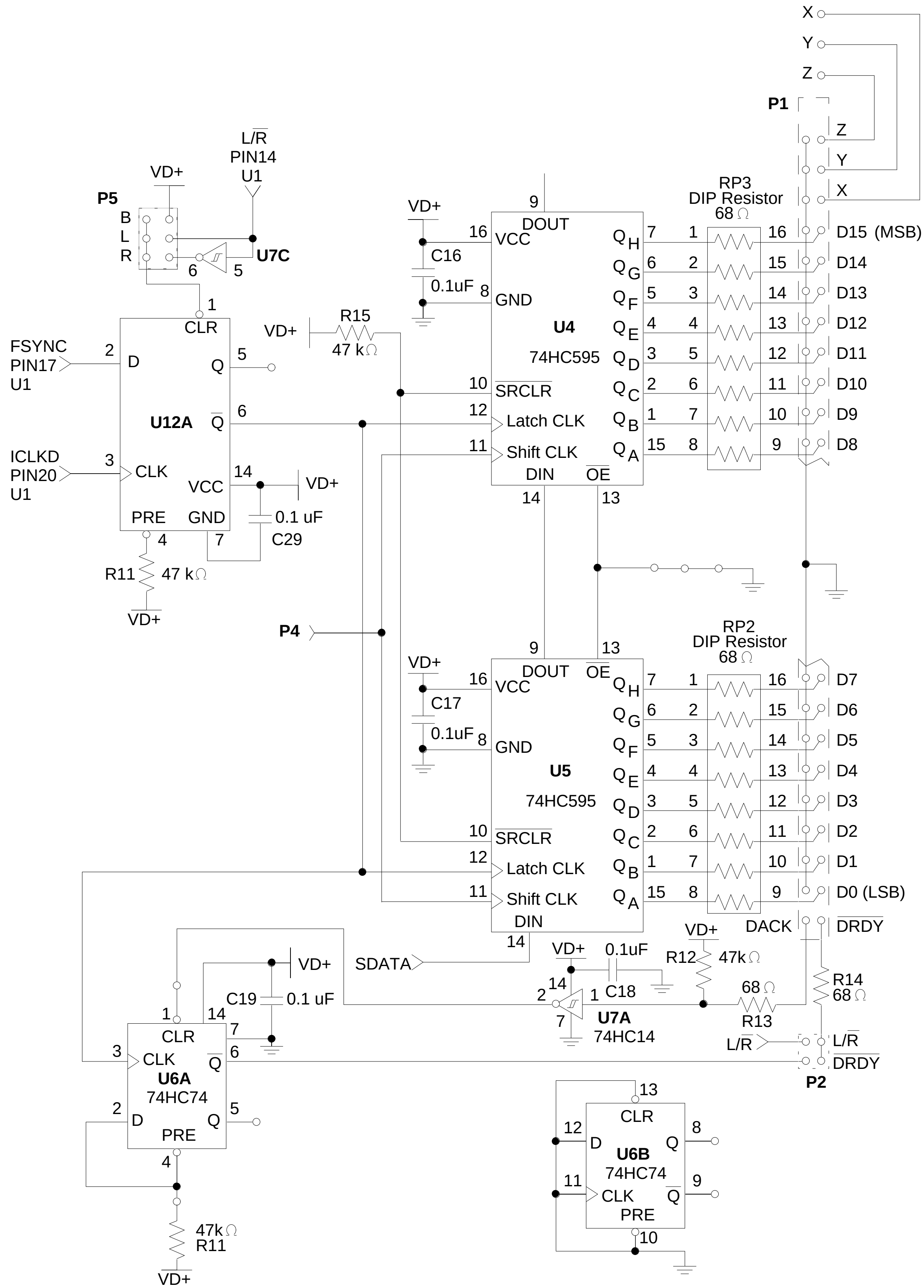


Figure 6. Parallel Output Interface

CONNECTOR	INPUT/OUTPUT	SIGNAL PRESENT
+15	input	+15 Volts from power supply
-15	input	-15 Volts from power supply
AGND	input	analog ground connection from power supply
+5	input	+5V for ADC VD+ and discrete logic
DGND	input	digital ground connection from power supply
AINL	input	left channel analog input
AINR	input	right channel analog input
EXTCLKIN	input	external master clock input
$L/\overline{R}$	output/input	left /right channel signal
SDATA	output	serial output data
SCLK	output/input	serial output clock
FSYNC	output/input	data framing signal
DIGITAL OUTPUT	output	CS8402 digital output via transformer
P3	output/input	CS8402 C,U,V inputs; CBL output
P1	output	parallel output data

Table 1. System Connections

JUMPER	PURPOSE	POSITION	FUNCTION SELECTED
P6	selects offset calibration input source	AIN	AINL and AINR selected during offset calibration
		*ZERO	ZEROL and ZEROR selected during offset calibration
P7	selects master clock source for CS5326 CLKIN	*INT	CLKIN provided by U3
		EXT	CLKIN provided by EXTCLKIN BNC
P5	selects channel for serial to parallel conversion	*L	left channel data presented on P1
		R	right channel data presented on P1
		B	left then right channel data alternately presented on P1
P2	selects $L/\overline{R}$ or $\overline{DRDY}$ as the output status signal presented on P1	*$\overline{DRDY}$	$\overline{DRDY}$ selected to signal the arrival of new data for the selected channel
		$L/\overline{R}$	$L/\overline{R}$ selected
P8, P9	selects optional input buffers	*IN	Buffer amplifier in circuit
		OUT	Buffer amplifier bypassed
P4	selects device type	5337/39	Correct SCLK for CS5337 & CS5339
		5336/38	Correct SCLK for CS5336 & CS5338

* Default setting from factory

Table 2. Jumper Selectable Options

Switch#	0=Closed, 1=Open	Comment
3	$\overline{\mathrm{PRO}}$=0	Professional Mode, C0=1 (default)
1	CRE	Local Sample Address Counter & Reliability Flags
default	0 1	Disabled Internally Generated (channel status bytes 14-17 and byte 22)
5, 2	$\overline{\mathrm{C6}}$, $\overline{\mathrm{C7}}$	C6,C7 - Sample Frequency
 default	1 1 1 0 0 1 0 0	00 - Not Indicated - Default to 48 kHz 01 - 48 kHz 10 - 44.1 kHz 11 - 32 kHz
4	$\overline{\mathrm{C1}}$	C1 - $\overline{\mathrm{Audio}}$
default	1 0	0 - Normal Audio 1 - Non-Audio
6	$\overline{\mathrm{C9}}$	C8,C9,C10,C11 - Channel Mode (1 of 4 bits)
 default	1 0	0000 - Not indicated - Default to 2-channel 0100 - Stereophonic
8, 7	EM1, EM0	C2,C3,C4 - Emphasis
default	1 1 1 0 0 1 0 0	000 - Not Indicated - default to none 100 - No emphasis 110 - 50/15 µs 111 - CCITT J.17

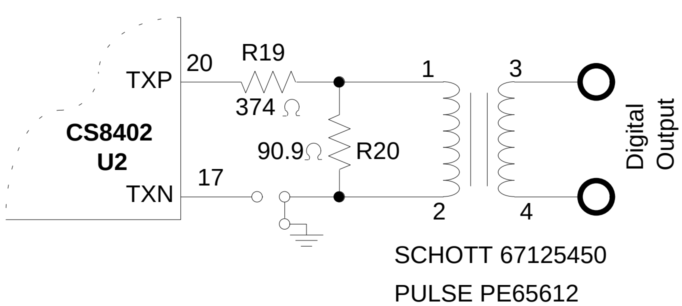
Table 3. CS8402 Switch Definitions - Professional Mode

and consumer modes; however, the CS8402 output to the transformer must be modified, as shown below Table 4, to be compatible with the consumer interface. Table 4 lists the switch settings for consumer mode. If the C input of connector P3 is used, the input bits are logically OR'ed with the appropriate DIP switch bits. In Tables 3 and 4, the 'C' bits listed in the comment section are taken from the Digital Audio Interface specifications. As an example, switch 6 in the professional mode (Table 3) controls $\overline{\mathrm{C9}}$ which is the inverse of channel status bit 9 (also listed as byte 1, bit 1 in the CS8402 data sheet). Channel status bit 9 is one of four bits indicating channel mode. Therefore, using DIP switch 6, only two of the available channel modes may be selected. The C input port on connector P3 may be used to select other channel modes. See the CS8401 & CS8402 part data sheet for more information on the operation of the CS8402.

Switch#	0=Closed, 1=Open	Comment
3	$\overline{\text{PRO}}$=1	Consumer Mode, C0=0 (Note 1)
1, 4	FC1, FC0	C24,C25,C26,C27 - Sample Frequency (encoded 2 of 4 bits)
	0 0 0 1 1 0 1 1	0000 - 44.1 kHz 0100 - 48 kHz 1100 - 32 kHz 0000 - 44.1 kHz, CD Mode
2	$\overline{\text{C3}}$	C3,C4,C5 - Emphasis (1 of 3 bits)
	1 0	000 - None 100 - 50/15 µs
5	$\overline{\text{C2}}$	C2 - Copy/Copyright
	1 0	0 - Copy Inhibited/Copyright Asserted 1 - Copy Permitted/Copyright Not Asserted
6	$\overline{\text{C15}}$	C15 - Generation Status
	1 0	0 - Definition is based on category code. 1 - See CS8402 Data Sheet, Appendix A
8, 7	$\overline{\text{C8}}$, $\overline{\text{C9}}$	C8-C14 - Category Code (2 of 7 bits)
	1 1 1 0 0 1 0 0	0000000 - General 0100000 - PCM encoder/decoder 1000000 - Compact Disk - CD 1100000 - Digital Audio Tape - DAT

Note: 1. The evaluation board is shipped from the factory in the Professional mode. Changing switch 3 to open places the CS8402 in Consumer mode; however, the hardware is not set up for consumer mode. To modify the hardware for Consumer mode, change R19 to 374Ω and add R20 at 90.9Ω. Then, as shown in the figure below, cut the trace connecting TXN to the transformer, and connect the transformer side to the ground hole provided. For a full explanation of the consumer hardware interface, see the CS8402 data sheet, Appendix B.

Table 4. CS8402 Switch Definitions - Consumer Mode



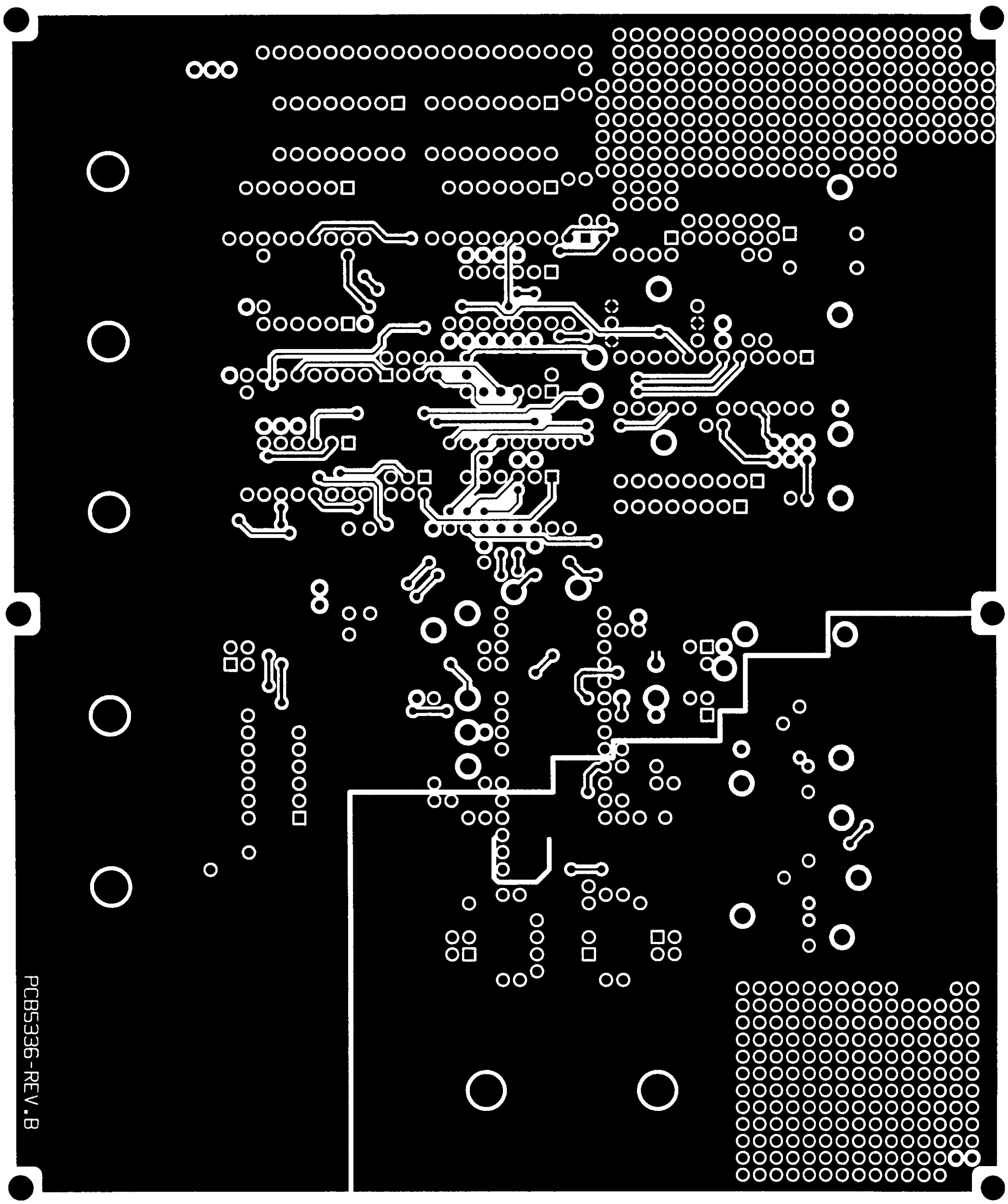


Figure 7. Top Ground Plane Layer (NOT TO SCALE)

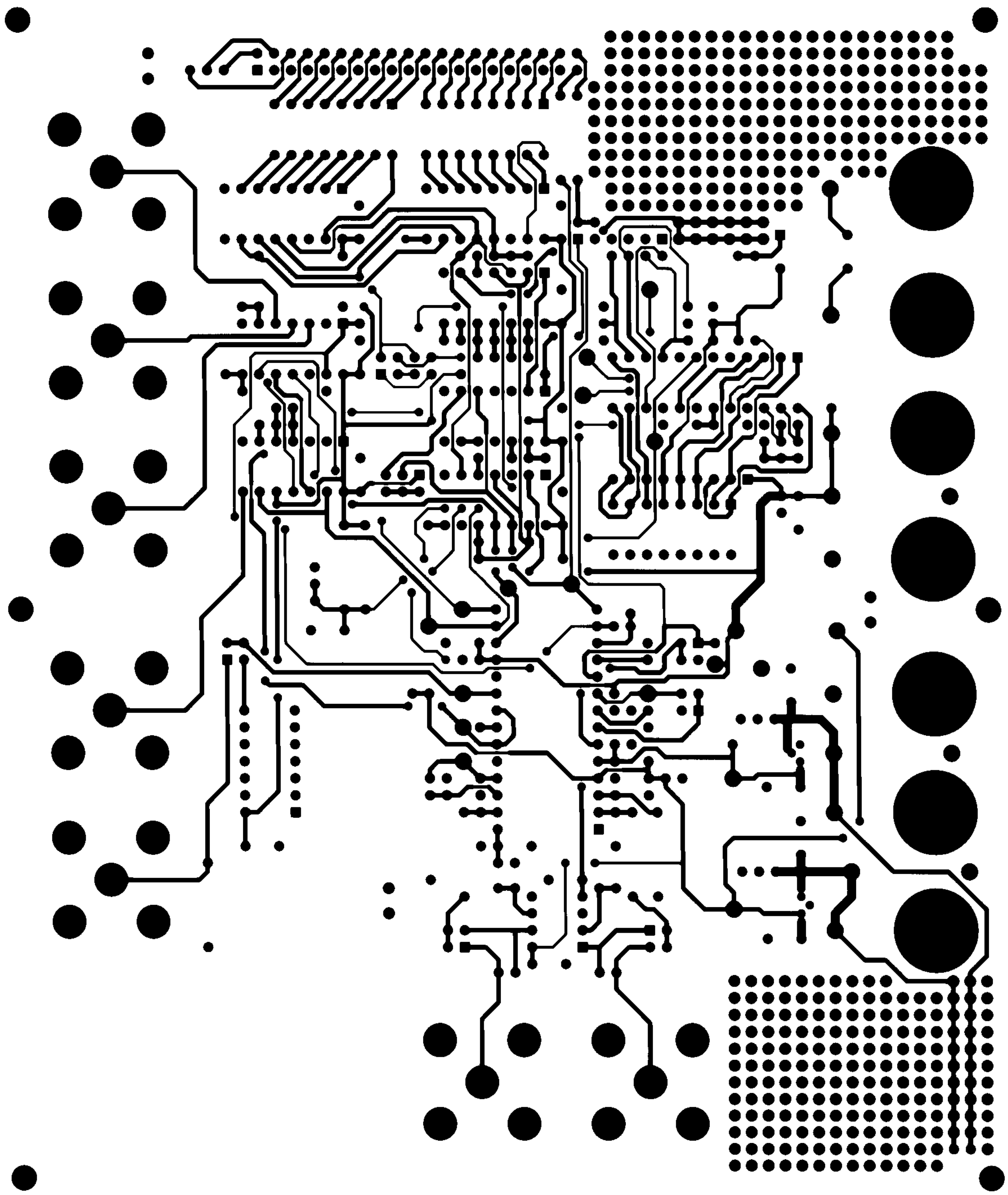


Figure 8. Bottom Trace Layer (NOT TO SCALE)

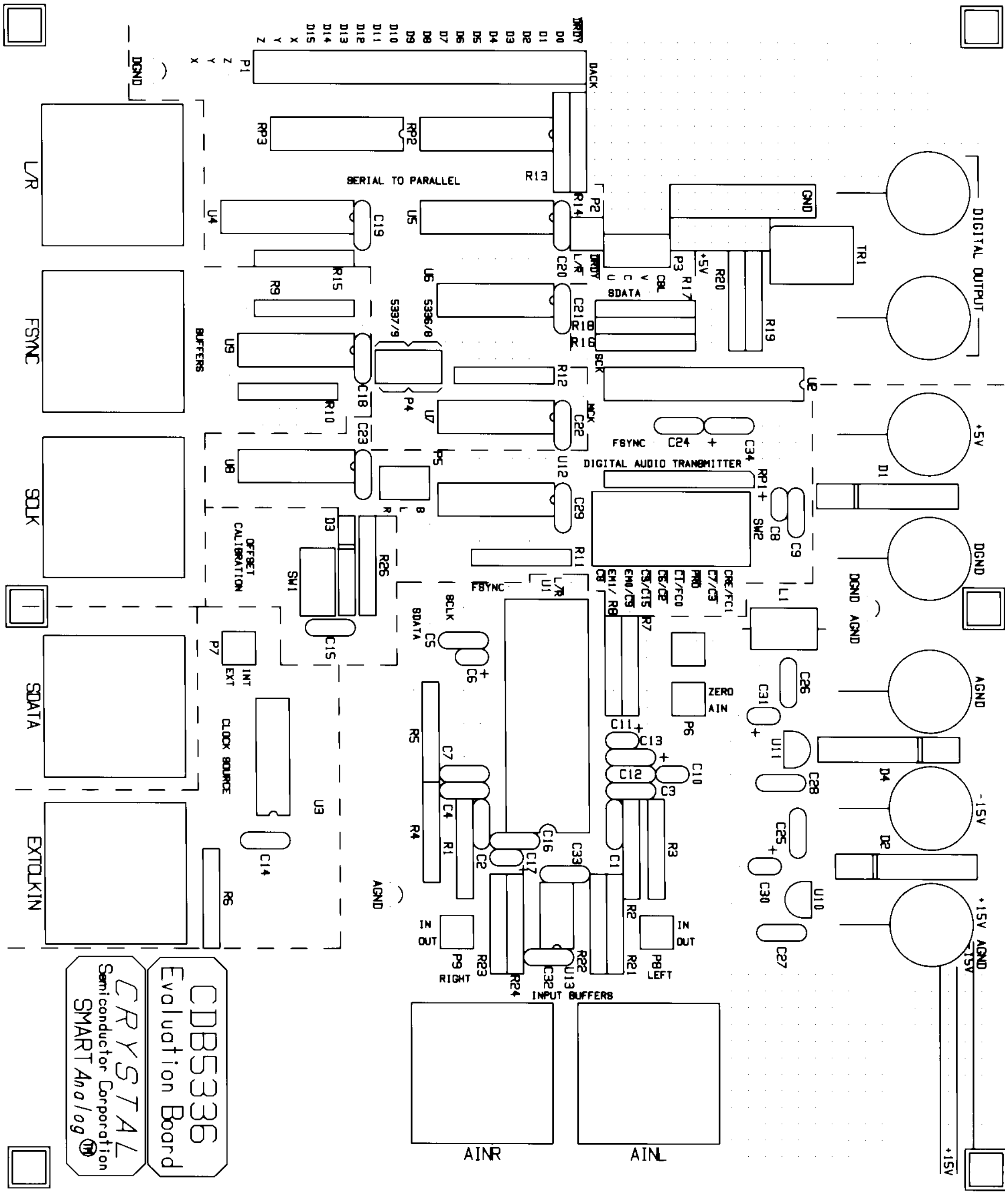


Figure 9. Component Layout (NOT TO SCALE)

PCM1702P
PCM1702U

BiCMOS Advanced Sign Magnitude 20-Bit DIGITAL-TO-ANALOG CONVERTER

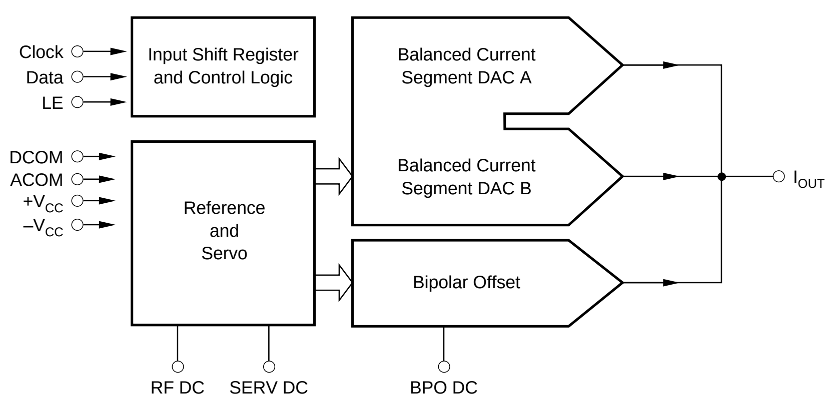
FEATURES

- ULTRA LOW –96dB max THD+N (No External Adjustment Required)
- NEAR-IDEAL LOW LEVEL OPERATION
- GLITCH-FREE OUTPUT
- 120dB SNR TYP (A-Weight Method)
- INDUSTRY STD SERIAL INPUT FORMAT
- FAST (200ns) CURRENT OUTPUT (±1.2mA)
- CAPABLE OF 16X OVERSAMPLING
- COMPLETE WITH REFERENCE
- LOW POWER (150mW typ)

DESCRIPTION

The PCM1702 is a precision 20-bit digital-to-analog converter with ultra-low distortion (–96dB typ with a full scale output). Incorporated into the PCM1702 is an advanced sign magnitude architecture that eliminates unwanted glitches and other nonlinearities around bipolar zero. The PCM1702 also features a very low noise (120dB typ SNR: A-weighted method) and fast settling current output (200ns typ, 1.2mA step) which is capable of 16X oversampling rates.

Applications include very low distortion frequency synthesis and high-end consumer and professional digital audio applications.



International Airport Industrial Park • Mailing Address: PO Box 11400 • Tucson, AZ 85734 • Street Address: 6730 S. Tucson Blvd. • Tucson, AZ 85706
Tel: (520) 746-1111 • Twx: 910-952-1111 • Cable: BBRCORP • Telex: 066-6491 • FAX: (520) 889-1510 • Immediate Product Info: (800) 548-6132

PDS-1175B

Printed in U.S.A. June, 1995

SBAS026

SPECIFICATIONS

All specifications at 25°C, $\pm V_{CC}$ and $+V_{DD}$ = ±5V unless otherwise noted.

PARAMETER	CONDITIONS	PCM1702P/U, -J, -K MIN	TYP	MAX	UNITS
RESOLUTION		20			Bits
DYNAMIC RANGE, THD + N at –60dB Referred to Full Scale, with A-weight			110		dB
DIGITAL INPUT					
Logic Family			TTL/CMOS Compatible		
Logic Level: V_{IH}		+2.4		$+V_{DD}$	V
V_{IL}		0		0.8	V
I_{IH}	$V_{IH} = +V_{DD}$			±10	µA
I_{IL}	V_{IL} = 0V			±10	µA
Data Format			Serial, MSB First, BTC(1)		
Input Clock Frequency			12.5	20.0	MHz
TOTAL HARMONIC DISTORTION + N(2)					
P/U V_O = 0dB	f_S = 352.8kHz(3), f = 1002Hz(4)		–92	–88	dB
V_O = –20dB	f_S = 352.8kHz(3), f = 1002Hz(4)		–82	–74	dB
V_O = –60dB	f_S = 352.8kHz(3), f = 1002Hz(4)		–46	–40	dB
P/U, -J V_O = 0dB	f_S = 352.8kHz(3), f = 1002Hz(4)		–96	–92	dB
V_O = –20dB	f_S = 352.8kHz(3), f = 1002Hz(4)		–83	–76	dB
V_O = –60dB	f_S = 352.8kHz(3), f = 1002Hz(4)		–48	–42	dB
P/U, -K V_O = 0dB	f_S = 352.8kHz(3), f = 1002Hz(4)		–100	–96	dB
V_O = –20dB	f_S = 352.8kHz(3), f = 1002Hz(4)		–84	–80	dB
V_O = –60dB	f_S = 352.8kHz(3), f = 1002Hz(4)		–50	–44	dB
ACCURACY					
Level Linearity	At –90dB Signal Level		±0.5		dB
Gain Error			±0.5	±3	%
Bipolar Zero Error(5)			±0.25		%
Gain Drift	0°C to 70°C		±25		ppm of FSR/°C
Bipolar Zero Drift	0°C to 70°C		±5		ppm of FSR/°C
Warm-up Time			1		minute
IDLE CHANNEL SNR(6)	Bipolar Zero, A-weighted Filter	110	120		dB
ANALOG OUTPUT					
Output Range			±1.2		mA
Output Impedance			1.0		kΩ
Settling Time	(±0.003% of FSR, 1.2mA Step)		200		ns
Glitch Energy			No Glitch Around Zero		
POWER SUPPLY REQUIREMENTS					
Supply Voltage Range: $+V_{CC} = +V_{DD}$		+4.75	+5.00	+5.25	V
$-V_{CC} = -V_{DD}$		–4.75	–5.00	–5.25	V
Combined Supply Current: $+I_{CC}$	$+V_{CC} = +V_{DD}$ = +5V		+5.00	+9.0	mA
Combined Supply Current: $-I_{CC}$	$-V_{CC} = -V_{DD}$ = –5V		–25.00	–41.0	mA
Power Dissipation	$\pm V_{CC} = \pm V_{DD}$ = ±5V		150	250	mW
TEMPERATURE RANGE					
Operating		–25		+85	°C
Storage		–55		+125	°C

NOTES: (1) Binary Two's Complement coding. (2) Ratio of ($Distortion_{RMS} + Noise_{RMS}$) / $Signal_{RMS}$. (3) D/A converter sample frequency (8 x 44.1kHz; 8x oversampling). (4) D/A converter output frequency (signal level). (5) Offset error at bipolar zero. (6) Measured using an OPA627 and 5kΩ feedback and an A-weighted filter.

ABSOLUTE MAXIMUM RATINGS (DIP Package)

Power Supply Voltage	±6.5VDC
Input Logic Voltage	DGND—0.3V~+V_{DD}+0.3V
Operating Temperature	–25°C to +85°C
Storage Temperature	–55°C to +125°C
Power Dissipation	500mW
Lead Temperature (soldering, 10s)	260°C

ABSOLUTE MAXIMUM RATINGS (SOP Package)

Power Supply Voltage	±6.5VDC
Input Logic Voltage	DGND—0.3V~+V_{DD}+0.3V
Operating Temperature	–25°C to +85°C
Storage Temperature	–55°C to +125°C
Power Dissipation	300mW
Lead Temperature (soldering, 5s)	260°C

PIN ASSIGNMENTS (DIP Package)

PIN	MNEMONIC	PIN	MNEMONIC
1	DATA	9	+V_{CC}
2	CLOCK	10	BPO DC
3	+V_{DD}	11	I_{OUT}
4	DCOM	12	ACOM
5	–V_{DD}	13	ACOM
6	LE	14	SERV DC
7	NC	15	REF DC
8	NC	16	–V_{CC}

PIN ASSIGNMENTS (SOP Package)

PIN	MNEMONIC	PIN	MNEMONIC
1	DATA	11	+V_{CC}
2	CLOCK	12	BPO DC
3	NC	13	NC
4	+V_{DD}	14	I_{OUT}
5	DCOM	15	ACOM
6	–V_{DD}	16	ACOM
7	LE	17	SERV DC
8	NC	18	NC
9	NC	19	RFE DC
10	NC	20	–V_{CC}

PACKAGE INFORMATION[1]

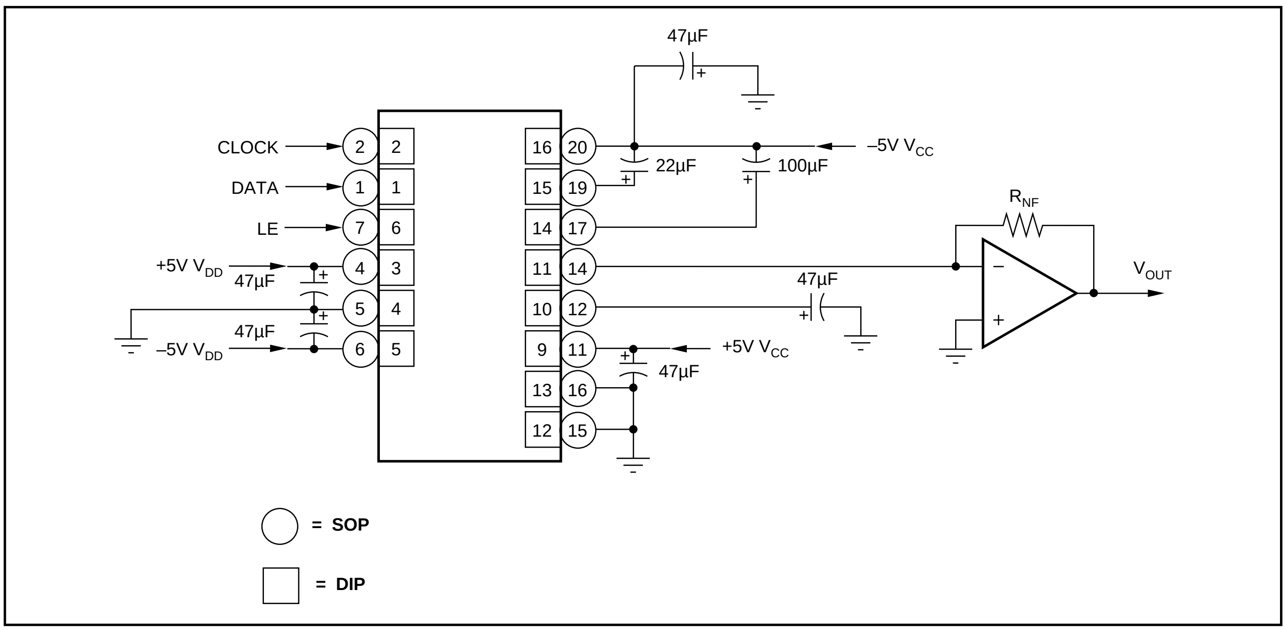
MODEL	PACKAGE	PACKAGE DRAWING NUMBER
PCM1702P	16-Pin Plastic DIP	180
PCM1702U	20-Pin Plastic SOP	248

NOTE: (1) For detailed drawing and dimension table, please see end of data sheet, or Appendix D of Burr-Brown IC Data Book.

GRADE MARKING (SOP Package)

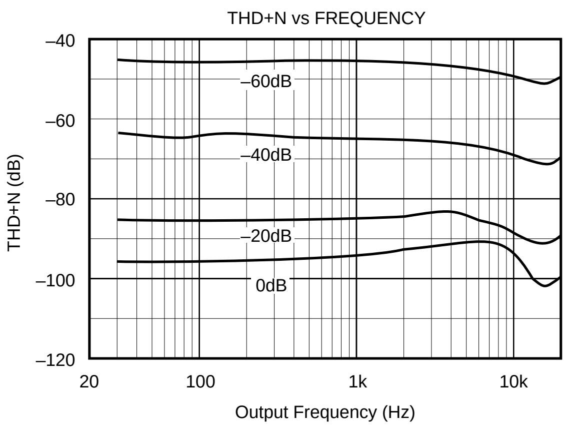
MODEL	PACKAGE
PCM1702U	Marked PCM1702.
PCM1702U-J	Marked with white dot by pin 10.
PCM1702U-K	Marked with red dot by pin 10.

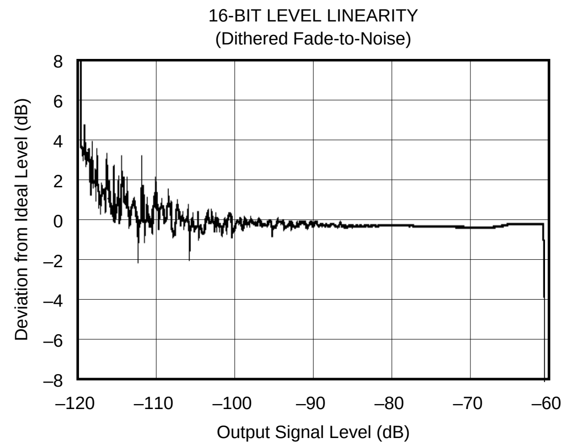
CONNECTION DIAGRAM

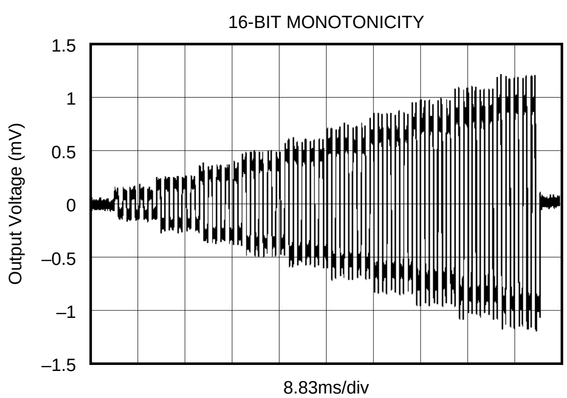


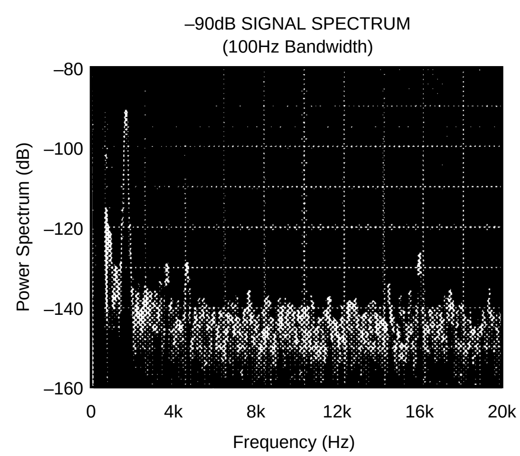
TYPICAL PERFORMANCE CURVES

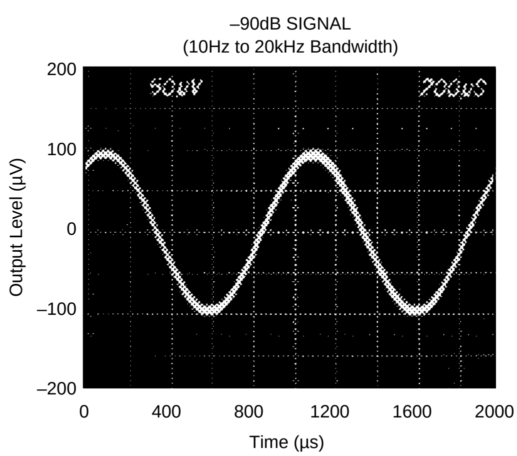
All specifications at 25°C, $\pm V_A$ and $\pm V_D$ = ±5.0V unless otherwise noted.

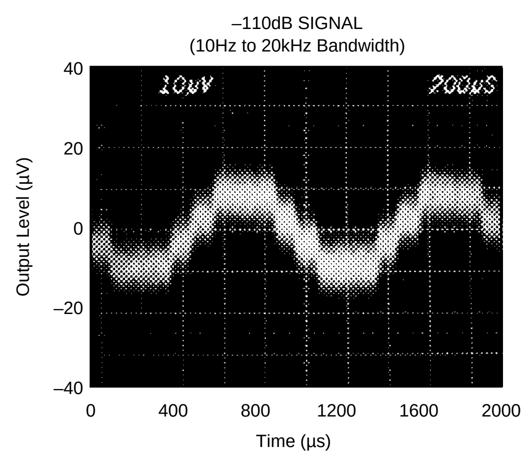












THEORY OF OPERATION

ADVANCED SIGN MAGNITUDE

Digital audio systems have traditionally used laser-trimmed, current-source DACs in order to achieve sufficient accuracy. However, even the best of these suffer from potential low-level nonlinearity due to errors at the major carry bipolar zero transition. More recently, DACs employing a different architecture which utilizes noise shaping techniques and very high over-sampling frequencies, have been introduced ("Bitstream", "MASH", or 1-bit DAC). These DACs overcome the low level linearity problem, but only at the expense of signal-to-noise performance, and often to the detriment of channel separation and intermodulation distortion if the succeeding circuitry is not carefully designed.

The PCM1702 is a new solution to the problem. It combines all the advantages of a conventional DAC (excellent full scale performance, high signal-to-noise ratio and ease of use) with superior low-level performance. Two DACs are combined in a complementary arrangement to produce an extremely linear output. The two DACs share a common reference, and a common R-2R ladder for bit current sources by dual balanced current segments to ensure perfect tracking under all conditions. By interleaving the individual bits of each DAC and employing precise laser trimming of resistors, the highly accurate match required between DACs is achieved.

This new, complementary linear or advanced sign magnitude approach, which steps away from zero with small steps in both directions, avoids any glitching or "large" linearity errors and provides an absolute current output. The low level performance of the PCM1702 is such that real 20-bit resolution can be realized, especially around the critical bipolar zero point.

Table 1 shows the conversion made by the internal logic of the PCM1702 from binary two's complement (BTC). Also, the resulting internal codes to the upper and lower DACs (see front page block diagram) are listed. Notice that only the LSB portions of either internal DAC are changing around bipolar zero. This accounts for the superlative performance of the PCM1702 in this area of operation.

DISCUSSION OF SPECIFICATIONS

DYNAMIC SPECIFICATIONS

Total Harmonic Distortion + Noise

The key specifications for the PCM1702 is total harmonic distortion plus noise (THD+N).

Digital data words are read into the PCM1702 at eight times the standard compact disk audio sampling frequency of 44.1kHz (352.8kHz) so that a sine wave output of 1002Hz is realized.

For production testing, the output of the DAC goes to an I to V converter, then through a 40kHz low pass filter, and then to a programmable gain amplifier to provide gain at lower signal output test levels before being fed into an analog-type distortion analyzer. Figure 1 shows a block diagram of the production THD+N test setup.

For the audio bandwidth, THD+N of the PCM1702 is essentially flat for all frequencies. The typical performance curve, "THD+N vs Frequency", shows four different output signal levels: 0dB, –20dB, –40dB, and –60dB. The test signals are derived from a special compact test disk (the CBS CD-1). It is interesting to note that the –20dB signal falls only about 10dB below the full scale signal instead of the expected 20dB. This is primarily due to the superior low level signal performance of the advanced sign magnitude architecture of the PCM1702.

In terms of signal measurement, THD+N is the ratio of $Distortion_{RMS} + Noise_{RMS} / Signal_{RMS}$ expressed in dB. For the PCM1702, THD+N is 100% tested at all three specified output levels using the test setup shown in Figure 1. It is significant to note that this test setup does not include any output deglitching circuitry. All specifications are achieved without the use of external deglitchers.

Dynamic Range

Dynamic range in audio converters is specified as the measure of THD+N at an effective output signal level of –60dB referred to 0dB. Resolution is commonly used as a theoretical measure of dynamic range, but it does not take into account the effects of distortion and noise at low signal levels. The advanced sign magnitude architecture of the PCM1702, with its ideal performance around bipolar zero, provides a more usable dynamic range, even using the strict audio definition, than any previously available D/A converter.

ANALOG OUTPUT	INPUT CODE (20-bit Binary Two's Complement)	LOWER DAC CODE (19-bit Straight Binary)	UPPER DAC CODE (19-bit Straight Binary)
+Full Scale	011...111	111...111+1LSB(1)	111...111
+Full Scale –1LSB	011...110	111...111+1LSB(1)	111...110
Bipolar Zero +2LSB	000...010	111...111+1LSB(1)	000...010
Bipolar Zero +1LSB	000...001	111...111+1LSB(1)	000...001
Bipolar Zero	000...000	111...111+1LSB(1)	000...000
Bipolar Zero –1LSB	111...111	111...111	000...000
Bipolar Zero –2LSB	111...110	111...110	000...000
–Full Scale +LSB	100...001	000...001	000...000
–Full Scale	100...000	000...000	000...000

NOTE: (1) The extra weight of 1LSB is added at this point to make the transfer function symmetrical around bipolar zero.

TABLE I. Binary Two's Complement to Sign Magnitude Conversion Chart.

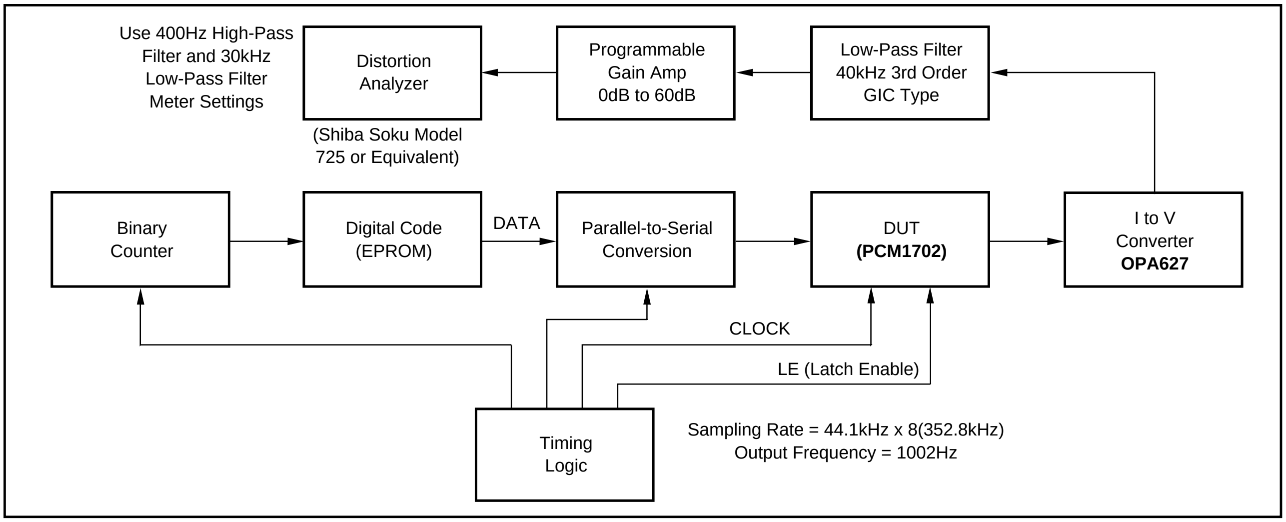


FIGURE 1. Production THD+N Test Setup.

Level Linearity

Deviation from ideal versus actual signal level is sometimes called "level linearity" in digital audio converter testing. See the "–90dB Signal Spectrum" plot in the Typical Performance Curves section for the power spectrum of a PCM1702 at a –90dB output level. (The "–90dB Signal" plot shows the actual –90dB output of the DAC). The deviation from ideal for PCM1702 at this signal level is typically less than ±0.3dB. For the "–110dB Signal" plot in the Typical Performance Curves section, true 20-bit digital code is used to generate a –110dB output signal.

This type of performance is possible only with the low-noise, near-theoretical performance around bipolar zero of the PCM1702 advanced sign magnitude.

A commonly tested digital audio parameter is the amount of deviation from ideal of a 1kHz signal when its amplitude is decreased form –60dB to –120dB. A digitally dithered input signal is applied to reach effective output levels of –120dB using only the available 16-bit code from a special compact disk test input. See the "16-bit Level Linearity" plot in the Typical Performance Curves section for the results of a PCM1702 tested using this 16-bit dithered fade-to-noise signal. Note the very small deviation from ideal as the signal goes from –60dB to –100dB.

DC SPECIFICATION

Idle Channel SNR

Another appropriate specification for a digital audio converter is idle channel signal-to-noise ratio (idle channel SNR). This is the ratio of noise on the DAC output at bipolar zero in relation to the full scale range of the DAC. To make this measurement, the digital input is continuously fed the code for bipolar zero, while the output of the DAC is band-limited from 20Hz to 20kHz and an A-weighted filter is applied. The idle channel SNR for the PCM1702 is typically greater than 120dB, making it ideal for low-noise applications.

Monotonicity

Because of the unique advanced sign magnitude architecture of the PCM1702, increasing values of digital input will always result in increasing values of DAC output as the signal moves away from bipolar zero in one-LSB steps (in either direction). The "16-bit Monotonicity" plot in the Typical Performance Curves section was generated using 16-bit digital code from a test compact disk. The test starts with 10 periods of bipolar zero. Next are 10 periods of alternating 1LSBs above and below zero, and then 10 periods of alternating 2LSBs above and below zero, and so on until 10LSBs above and below zero are reached. The signal pattern then begins again at bipolar zero.

With PCM1702, the low-noise steps are clearly defined and increase in near-perfect proportion. This performance is achieved without any external adjustments. By contrast, sigma-delta ("Bit-stream", "MASH", or 1-bit DAC) architectures are too noisy to even see the first 3 or 4 bits change (at 16 bits), other than by a change in the noise level.

Absolute Linearity

Even though absolute integral and differential linearity specs are not given for the PCM1702, the extremely low THD+N performance is typically indicative of 17-bit integral linearity in the DAC. The relationship between THD+N and linearity, however, is not such that an absolute linearity specification for every individual output code can be guaranteed.

Offset, Gain, and Temperature Drift

Although the PCM1702 is primarily meant for use in dynamic applications, specifications are also given for more traditional DC parameters such as gain error, bipolar zero offset error, and temperature gain and offset drift.

DIGITAL INPUT

Timing Considerations

The PCM1702 accepts TTL compatible logic input levels. The data format of the PCM1702 is binary two's complement (BTC) with the most significant bit (MSB) being first in the serial input bit stream. Table II describes the exact relationship of input data to voltage output coding. Any number of bits can precede the 20 bits to be loaded, since only the last 20 will be transferred to the parallel DAC register after Latch Enable (Pin6 <PCM1702P>, Pin7 <PCM1702U>, LE) has gone low.

All DAC serial input data (Pin1, DATA) bit transfers are triggered on positive clock (Pin2, CLOCK), edges. The serial-to-parallel data transfer to the DAC occurs on the falling edge of Latch Enable. The change in the output of the DAC occurs at a rising edge of the 4th clock of the CLOCK after the falling edge of Latch Enable. Refer to Figure 2 for graphical relationships of these signals.

Maximum Clock Rate

A typical clock rate of 16.9MHz for the PCM1702 is derived by multiplying the standard audio sample rate of 44.1kHz by sixteen times (16X over-sampling) the standard audio word bit length of 24 bits (44.1kHz x 16 x 24 = 16.9MHz). Note that this clock rate accommodates a 24-bit word length, even though only 20 bits are actually being used. The setup and hold timing relationships are shown in Figure 3.

"Stopped Clock" Operation

The PCM1702 is normally operated with a continuous clock input signal. If the clock is to be stopped between input data words, the last 20 bits shifted in are not actually shifted from the serial register to the latched parallel DAC register until Latch Enable goes low. Latch Enable must remain low until after the first clock cycle of the next data word to insure proper DAC operation. In any case, the setup and hold times for Data and LE must be observed as shown in Figure 3.

DIGITAL INPUT	ANALOG OUTPUT	CURRENT OUTPUT
1,048,576LSBs	Full Scale Range	2.40000000mA
1LSB	NA	2.28882054nA
$7FFFF_{HEX}$	+Full Scale	–1.19999771mA
00000_{HEX}	Bipolar Zero –1LSB	0.00000000mA
80000_{HEX}	–Full Scale	+1.20000000mA

TABLE II. Digital Input/Output Relationships.

INSTALLATION

POWER SUPPLIES

Refer to CONNECTION DIAGRAM for proper connection of the PCM1702. The PCM1702 only requires a ±5V supply. Both positive supplies should be tied together at a single point. Similarly, both negative supplies should be connected together. No real advantage is gained by using separate analog and digital supplies. It is more important that both these supplies be as "clean" as possible to reduce coupling of supply noise to the output. Power supply decoupling capacitors should be used at each supply pin to maximize power supply rejection, as shown in CONNECTION DIAGRAM regardless of how good the supplies are. Both commons should be connected to an analog ground plane as close to the PCM1702 as possible.

FILTER CAPACITOR REQUIREMENTS

As shown in CONNECTION DIAGRAM, various size decoupling capacitors can be used, with no special tolerances being required. The size of the offset decoupling capacitor is not critical either, with larger values (up to 100µF) giving slightly better SNR readings. All capacitors should be as close to the appropriate pins of the PCM1702 as possible to reduce noise pickup from surrounding circuitry.

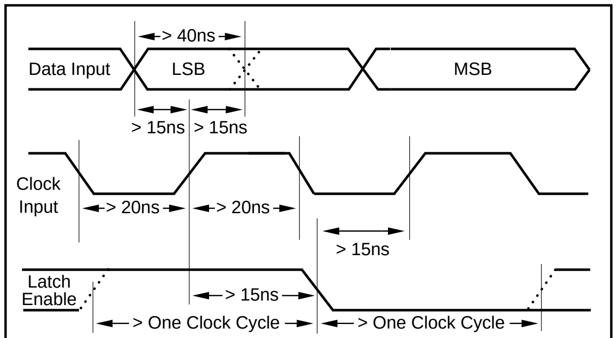
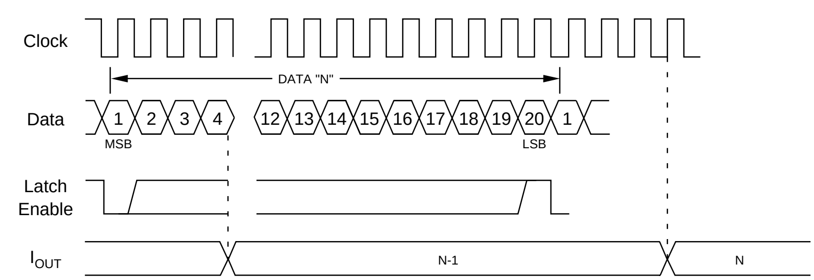


FIGURE 3. Setup and Hold Timing Diagram.



NOTES : (1) If clock is stopped between input of 20-bit data words, "Latch" Enable (LE) must remain low until after the first clock cycle of the next 20-bit data word stream. (2) Data format is binary two's complement (BTC). Individual data bits are clocked in on the corresponding positive clock edge. (3) Latch Enable (LE) must remain low at least one clock cycle after going negative. (4) Latch Enable (LE) must be high for at least one clock cycle before going negative. (5) I_{OUT} changes on positive going edge of the 4th clock after negative going edge of Latch Enable (LE).

FIGURE 2. Timing Diagram.

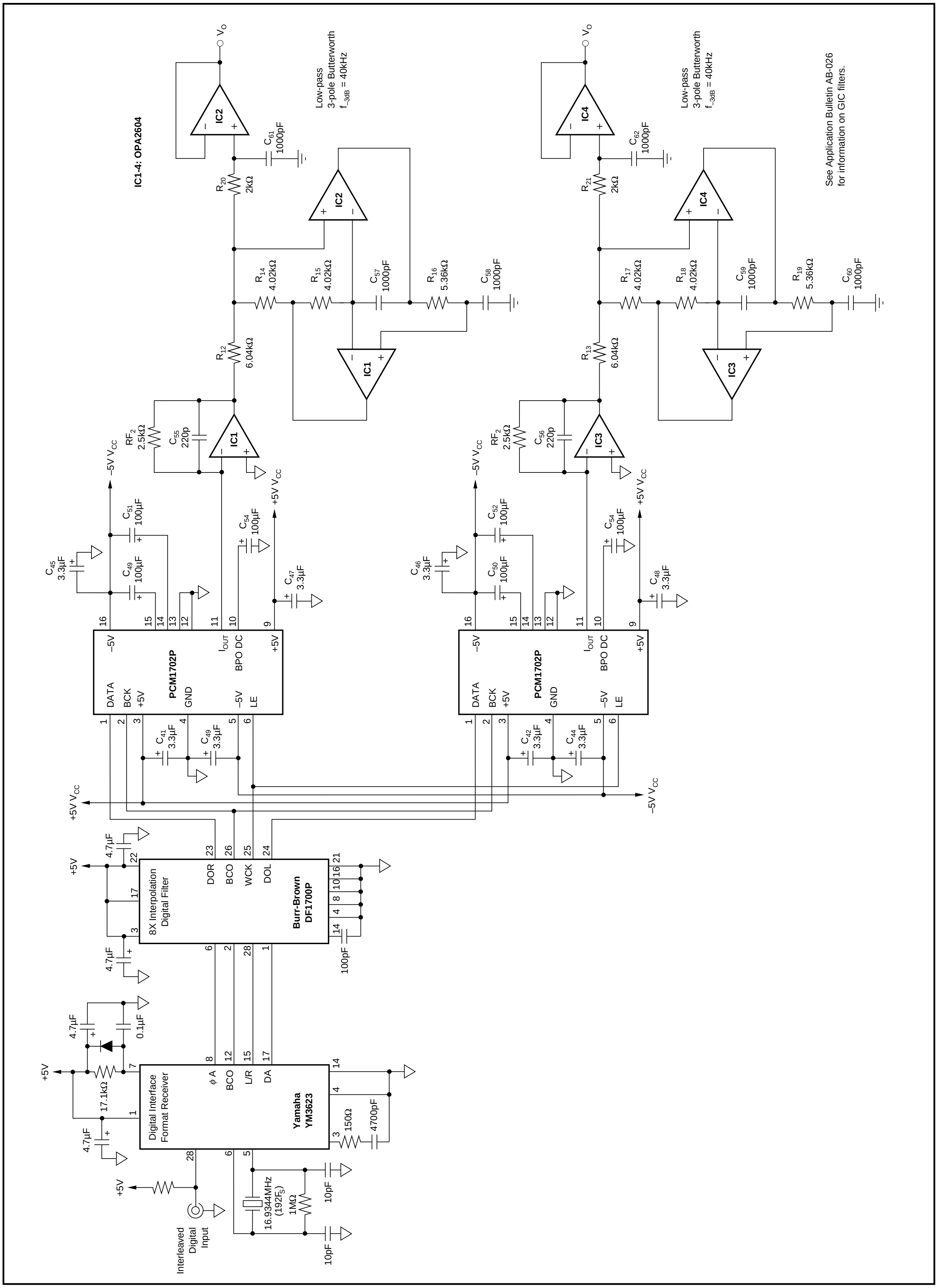


FIGURE 4. Typical Application for Stereo Audio 8X Oversampling system.

IMPORTANT NOTICE

Texas Instruments and its subsidiaries (TI) reserve the right to make changes to their products or to discontinue any product or service without notice, and advise customers to obtain the latest version of relevant information to verify, before placing orders, that information being relied on is current and complete. All products are sold subject to the terms and conditions of sale supplied at the time of order acknowledgment, including those pertaining to warranty, patent infringement, and limitation of liability.

TI warrants performance of its semiconductor products to the specifications applicable at the time of sale in accordance with TI's standard warranty. Testing and other quality control techniques are utilized to the extent TI deems necessary to support this warranty. Specific testing of all parameters of each device is not necessarily performed, except those mandated by government requirements.

Customers are responsible for their applications using TI components.

In order to minimize risks associated with the customer's applications, adequate design and operating safeguards must be provided by the customer to minimize inherent or procedural hazards.

TI assumes no liability for applications assistance or customer product design. TI does not warrant or represent that any license, either express or implied, is granted under any patent right, copyright, mask work right, or other intellectual property right of TI covering or relating to any combination, machine, or process in which such semiconductor products or services might be or are used. TI's publication of information regarding any third party's products or services does not constitute TI's approval, warranty or endorsement thereof.