MERIDIAN 551 USER MANUAL

MERIDIAN 601 USER MANUAL

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Designers and consultants

The 601 design team included Phil Boddy, Andrew Dutton, Richard Hollinshead, Andrew (Boris) Samaltonis, Bob Stuart and Rhonda Wilson.

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Author: J R Stuart

Introduction

The Meridian 601 DSP Preamplifier is part of the 600-Series of advanced high-fidelity equipment.

Because the 601 is such a sophisticated device, you are advised to read through this whole manual before using it for the first time, particularly if you are using it with other Meridian components. We have given a lot of attention to making the operation of the unit as intuitive as possible, so that those of you who are familiar with high fidelity equipment and who cannot wait to get your 601 working will be able to get immediate results. A 'Getting going' sheet has also been provided.

The Meridian 601 DSP Preamplifier is part of the 600-Series of advanced high fidelity equipment. Used in conjunction with other components in the range, it can offer the music lover much more than just very high quality audio.

In purchasing a Meridian 601 DSP Preamplifier, you have acquired a refined piece of equipment that will continue to bring you listening pleasure for many years. This manual will enable you to get the most from it.

DSP is short for Digital Signal Processing, a relatively modern technique where a specialised computer is used to calculate modifications to an audio signal. In 601 DSP replaces many of the functions previously made in analogue circuits – and also provides the possibility for many that are nearly impossible without DSP. A very important benefit of DSP is that, if the programmes are carefully written, the processing can be transparent, giving for the first time the possibility of useful features with full high sound quality.

About this manual

602 A number such as 602 refers to a component in the Meridian 600-Series of audio equipment.

Note. Whenever a key name is printed in bold typeface, it means that you should press this key. If the key name is not in bold type, it indicates that the course of action is not required at present, e.g. 'if you want to change the display, press Display' means you can do this, but it is not essential as part of the operation currently being discussed.

Standby Italic form normally refers to the state of the 601. In some instances, a key and the status of the equipment share the same name, e.g. the Standby key and the *Standby* state.

Loudness DSP programme names are written in bold italic.

Radio 54	Displays are illustrated in this form.

1. In this book, sequences of actions are given in numbered lists.

Note Some paragraphs may be in the form of notes, warnings or tips; these sections are delineated like this one. We recommend you do not skip these.

About the 600-Series

The *Meridian* 600-Series of high fidelity equipment is probably the most sophisticated family of hi-fi products available. It has been developed to operate at a variety of levels, from the simplest conventional single-room hi-fi to a whole-house music-management system. The 600-Series is also fully compatible with the famous 200-Series. In addition, the system can be further enhanced by the use of one or more pairs of *Meridian* D600 or D6000 Digital Loudspeakers.

The 601 DSP Preamplifier is very advanced. Its circuitry has been carefully optimised to provide the highest quality sound.

Despite its high performance, each piece of equipment in the 600-Series has been planned for ease of use, with microprocessor control being applied in the range to simplify operation. We have also paid careful attention to the appearance of the units, and great care has been taken in the selection of the materials used in the construction of the components. Your *Meridian* 600-Series equipment will look and sound as good in the years to come as it does now.

Unpacking the 601

On opening the carton, you should have found the following parts:

- Meridian 601 DSP Preamplifier
- Meridian 601 Remote Control
- Power cord
- This manual
- Getting going instruction sheet
- 4mm banana plug for turntable grounding

If any of these items is missing, please contact your dealer. We suggest that you retain the packing carefully as it provides maximum protection for the unit in transit.

Installing your 601

General precautions

Before carrying out any installation, you should ensure that the 601 is marked with the correct voltage for your local AC supply. Should this not be the case, do not proceed – but contact your dealer.

As a general rule, you should not make any connections to the 601, or to any other component in your system, whilst the AC power supply is connected and switched on.

Customising features

For full details of Type settings and configuring 601, see 'Customising 601, an overview' on page 18.

The 601 is very flexible. You have the opportunity to significantly customise its functions. In the end this gives you a preamplifier that exactly suits you.

The section 'Customising 601, an overview' on page 18 describes the options in detail. Obviously, when you have set up 601 to your needs, you will be aware of the differences from the standard configurations.

We recommend that you first get to know the 601 using one of the standard setups, and by working through this manual. Do not be afraid later to experiment with customising – we have given you a simple way to get back to where you started! (See page 24).

601 as provided has six standard configurations that we call *Type* settings. Each *Type* gives a different set of options for the two outputs; this feature is described fully on page 24. There are many more than five possible configurations for 601, but these five were chosen to give good starting points for customising.

By selecting a *Type* you automatically reset all custom settings for 601 to that default including DSP menu options.

601 has five operating modes (not to be confused with setup *Types*):

- *Normal*
- *Standby*
- *Type*
- *Config*
- *Setup*

Normal and *Standby* are the everyday operating modes: the basic operating instructions refer to these.

Type, *Config* and *Setup* are used to make choices about the way 601 works. These are described later in the section: 'Configuring 601, an overview' on page 18.

Connections

For details of making all connections, see Connecting 601, page 32.

There are three types of connection that need to be made to the 601:

- analogue audio
- digital audio
- communications.

Audio connections need to be made to other components in your system. Communication connections have to be made to most of the other Meridian components in your system in order for them to act as one system. The connections to your 601 are very important and care should be taken in deciding which connections to make.

Connection details for a wide variety of systems are described in the sections following page 32.

Siting

We recommend the following guidelines for location.

- The 601 should not be placed on top of a power amplifier.
- If you intend to use the moving coil LP input keep the left-hand side of the 601 as far as possible from other mains-operated equipment. See *help item* on page 72.
- Avoid sites where it would be subjected to direct sunlight or where it is near a heat source, e.g. a central heating radiator.
- Avoid sites where the 601 could be subjected to strong magnetic radiation, like near a power amplifier.
- Not too close to a television, or where connecting cables may be subjected to, or cause, interference.

Tip: In common with all digital products, the 601 emits some radio-frequency signals. For this reason, try to keep all power cables and network leads away from audio, antenna and loudspeaker cables. Some time spent laying out the cables carefully will repay you with the best possible performance later.

Starting off with 601

Checking

Before turning on the power, check once again that you have made all the connections correctly and that you have not disturbed any existing connections in the process.

Switching on

The power switch is located directly next to the power inlet on the right-rear of 601.

1. Turn on the power switch at the rear of the unit. A small point on the display should now be illuminated and the legends lit up.

If there are no lights

Check the integrity of your power connections, including any fuses in your supply. If the standby point is still not illuminated and the rest of your system is functioning, contact your dealer for help.

Standby

The entire 600-Series is designed to be left connected to AC power at all times.

1. This standby state ensures that the components operate at maximum efficiency from the moment you start listening. It is perfectly safe and consumes a negligible amount of power. However, when you are not going to use your system for an extended period of time, e.g. when you are going on holiday, we would advise you to disconnect it from the AC power supply.

Basic operation of 601

Note 601 has an operating mode specifically for use with Meridian digital loudspeakers like D600 or D6000. The section 'Using D600 or D6000 speakers with 601' on page 55, describes how the operation differs from the following description.

Selecting a source

1. press **Source** until the input you require is indicated
2. release **Source**

   Selecting a source will bring 601 out of standby.

Putting the 601 in standby

1. press **Off** on the 601

   or

   press **Standby** – the only green key on the Remote Control

Coming out of standby

To start up the 601 from standby:

1. Select a source from the front panel or using the Remote Control. Tapping **Source** once brings the 601 on with the last-used source selected.

   or

   If the 601 is part of a *Meridian* system, bring any component out of standby. For example start up a *Meridian* CD player.

Adjusting the volume

601 has two outputs; 1 and 2. Normally the first output volume is displayed on 601.

You can adjust this volume on the front panel of the 601 or from the Remote Control.

To increase the volume

1. press ∧, the right-hand volume key on the 601

   or

   press ∧, the top red key on the Remote

To decrease the volume

1. press ∨, the left hand **Volume** key on the 601

   or

   press ∨, the bottom red key on the Remote Control

As you adjust the volume setting, the display will change to show the current source and volume level. For example:

CD	65

would indicate that you were listening to a Compact Disc at a volume setting of 65. The volume level displayed will count up or down and ranges from 1 to 99. Each step is 1dB.

If you have selected a different display, the display will then revert to your selected information.

Subjectively a volume increase of 9dB is equivalent to a doubling of loudness. Each volume number represents about a 11% change in loudness; nine steps to double loudness. In a traditional preamplifier using a rotary volume control the mid-way position offered between 20dB and 30dB of attenuation. On 601 20dB attenuation is at volume number 79, 30dB at 69. For normal to high-level listening you should expect volume numbers in the region 60 to 90. For this reason 601 starts at power-up with volume number 70.

Note When 601 is set up for use with D600 or D6000; first volume, balance and mute are handled by the speaker and 601 will not respond to the Remote control.

Note If you have selected the output to be Digital Variable instead of Analogue Variable then the display will be different. See page 66.

Adjusting the second volume

601 has two outputs; 1 and 2. Normally the first output volume is displayed on 601.

You can adjust the second volume on the front panel of the 601 or from the Remote Control if it is configured to be variable.

To adjust the second volume from the front panel:

1. Press Display until the display reads:

2nd 76

2. use the ∧ or ∨ **Volume** keys in the same way as for the first output.

Note If the first output is configured as 'fixed' then the front panel ∧ or ∨ Volume keys will adjust the second output – unless it is also 'fixed'. See pages 66 and 19.

To adjust the second volume from the Remote Control

1. Use the **Yellow** keys on the Remote. (This function is not available if the first output is fixed and D6000 mode is selected in Config, p. 23).

Muting the sound

The mute facility cuts off the sound by setting the volume level to zero.

Note In the same way as for the volume control, Mute operates on the output being displayed. The only exception to this is if the first and second outputs are ganged together – like in a surround system – or if one of the outputs is set up to be fixed, not variable. For more details see 'How mute works in different configurations' on page 67.

To mute the sound on the first output

1. Press **Mute** on the 601 or the Remote Control. The output will mute and the display will change to:

Muted

To restore the sound

1. Press **Mute** a second time, either on the 601 or the Remote Control. The volume level and display will be restored.

   or

   Use the **Volume** keys on either the 601 or Remote Control.

   Volume setting 41 will be selected and then normal volume control function returns at this low level.

Note When 601 is set up for use with D600 or D6000, find volume, balance, phase and mute are handled by the speaker and 601 will not respond to the Remote control.

To mute the sound on the second output

1. Press **Display** until the display reads:

2nd 76

2. press **Mute** on either the 601 or the Remote Control. The display will change to:

2nd MU

   and the output will be muted.

To restore the sound

1. Press the **Mute** key on either the 601 or the Remote Control a second time. The volume level will revert back to normal and the display will be restored.
   or
   when the display reads
   `2nd Mfr`
   use either the **Volume** keys on the 601 or the **yellow keys** on the Remote control.
   Volume setting 40 will be selected and then normal volume control returns.

Changing the display

The 601 has several display modes. To change the display:

1. Press **Display**.
   The 601 will cycle through the displays you have selected in setup. The standard displays are:
   - Source and first volume number (or gain)
   - Second volume number – if available
   - Display and legends off
   In addition, for CD sources the 601 displays:
   - Track and index numbers
   - Track time
   - Total time

The track number display will consist of one or two digits with the index point given after a decimal point, for example a display that reads:

`CD 11.3` would indicate that track 11, index 3 was currently being played, whereas:

`CD 3.42` would be a time display. When a CD has ended, the display will show a null display like:

`CD -:- -` When the selected source is not assigned one of the CD logos and is not digital or does not contain subcode, the track and time displays are omitted.

Note When 601 is set up for use with D600 or D6000 the 601 display can only be changed on its front panel. The Display key on the Remote control will control the speaker display.

Changing the DSP function

DSP stands for Digital Signal Processing. At the heart of 601 is a very powerful general purpose audio computer (Motorola 56001: 32MHz). By fitting a general purpose computer, we are able to provide a wide range of programmes to run on it – now, and in the future. In just the same way as a personal computer can run many types of programme just by loading it from a disc, 601 has been designed to have upgradable DSP programmes. In this case the programs are stored in a plug-in memory chip called an EPROM.

As supplied 601 has eight user programs that provide a wide range of facilities. These DSP modes are described in detail in the DSP section starting on page 38.

To change the DSP programme:

1. Press **DSP** on either the front panel or on the Remote Control. The first press will review the name of the current programme.

   Subsequent presses cycle through the options in the EPROM.

   or

   Press and hold the **DSP** key. 601 cycles through the available selections.

2. Release the **DSP** key. 601 will start the new DSP programme with the message:

Loading

You can review the DSP programme in standby. With the 601 in standby (after pressing Off):

1. Press **DSP** once – on the front panel or the Remote control.

   The display will show the name and version number of the current DSP programme.

To review DSP options in standby:

1. Press and hold **DSP**.

   The display will show in turn the names and version numbers of the available DSP programmes.

To change the DSP in standby:

1. Press and hold DSP.

   The display will show in turn the names and version numbers of the available DSP programmes. When you see the programme you want:

2. Release DSP.

   The 601 will display:

Loading

Other information reviewed in standby:

1. Press and hold Display.

   The display will show the version number of the microcontroller software, e.g.:

Ver 1.00

Advanced operation using the Remote Control

See section 'Balance control with and without delay' on page 68.

Balance control

The balance can only be adjusted from the Remote control.

Note When 601 is set up for use with D600 or D6000, volume, balance, phase and mute are handled by the speaker and 601 will not respond to the Remote control.

To move the sound to the left

1. Press < **Left**

The display will show the current balance, e.g.

← 1

The number indicates the number of dBs advantage given to the left channel.

To move the balance back again

1. Press > **Right** until the display says:

← 0→

or

Press **Reset** (also only on the Remote control)

Reset does note only restore the balance to central, it restores other settings depending on the DSP programme being run. See the description for the DSP you are using – following page 38.

To move the sound to the right

1. Press > **Right**

The display will show the current balance, e.g.

0→

The number indicates the number of dBs advantage given to the right channel.

Phase

The **Phase** key on the Remote Control inverts the absolute phase of the signal. To review the absolute phase:

1. Press **Phase**

The display will respond with:

Phase +

or

Phase –

To change the absolute phase

1. Press **Phase** again.

Note When 601 is set up for use with D600 or D6000, volume, balance, phase and mute are handled by the speaker and 601 will not respond to the Remote control.

The Menu system

The DSP programmes include a very large number of choices. Rather than clutter the 601 front panel or the Remote control with hundreds of switches you may never want to use, we have provided a flexible menu system to guide you through the options. One important advantage this gives is complete flexibility for future DSP options. *Please note:*

- The menus are different in each DSP programme.
- Menus are also used in *Config* and *Setup* modes.
- The DSP menus are also different in the Normal and Setup modes. See the section 'Customising using Setup' on page 28.
- Menus can only be accessed from the Remote control.

To access the menus

1. Use the **Blue** keys on the Remote control.

In most circumstances there is more than one menu available. For example the DSP programme 'Octave Equaliser' has one menu for each of the ten octave bands between 32Hz and 16kHz.

To move between menus

1. Press either < or > **blue** key on the Remote control.

Tip This is also the way to review settings you have made. The 601 will cycle through its menus, displaying the current choice in **each** one as it goes.

To change the choice in a menu

1. Press either ∧ or ∨ **blue** key on the Remote control.

For the example of DSP programme 'Octave Equaliser', these keys allow you to adjust the octave response ± 7dB in 1dB steps.

Switching the DSP on or off, Filter key

In most of the DSP programmes, the audio computer will be making changes to the signal under your direction from the menu choices. In addition the 601 will always use the computer to do the following things:

- Volume control and muting
- Output computing: Stereo, Mono, Mono L or Hafler™
- Equalisation for LP sources
- Monitoring for overload.

Usually the choices you make from the menus involve the use of digital filtering. You may want to switch the filtering on or off, this is a good way to review the settings you have made.

To review the filter setting

1. Press the **Filter** key on the Remote control.
   The 601 will display the filter status like:

Filt Off

To change the filtering

1. While the status is being displayed:
2. Press the **Filter** key on the Remote control again.
   The filter status will then toggle between 'On' and 'Off'.

Note *Bypass* does not have any filtering so there will be no response from the 601 display if this is the current DSP programme. In *Archive* you cannot switch the filtering off.

Memory

The 601 can store in permanent memory your choices of menu settings for each DSP programme. In normal use the **Memory** key will only recall the settings you have chosen to store. We arranged it this way so that the settings you have chosen – maybe based on hours of listening – are not lost if another user changes them. Menu settings can only be stored in Setup mode.

When you change DSP programmes, the stored settings for that programme are automatically loaded.

To recall the stored settings

1. Press **Memory**

   The 601 responds with the display message:

Recall

To reset the menu settings to the factory default for the current DSP programme:

1. Press **Reset**

Tip: Memory Recall and Reset allow you to compare complete combinations of settings directly.

To store settings in memory

To store settings in permanent memory the 601 must be in Setup mode. The method for entering Setup mode is described on page 31. If you have already chosen settings you want to put in the memory, use the review feature:

1. Press either < or > blue key on the Remote control.
2. Write down each setting – we have given you a place for that on page 79.
3. turn to page 31.

Reset

Reset restores:

- factory default settings for the DSP menus
- balance to central

To restore settings:

1. Press **Reset**, 601 responds with the display:

Reset

Reset does note only restore the menu settings, it also sets balance to central.

Controlling the CD player

If you have a Meridian CD player, you can control its basic functions from the 601 remote control. This allows you to keep only one remote controller to hand.

Note If you have an installation including D600 or D6000 then the speaker takes control from the 601. In these installations the 601 remote control works but the controlling displays are on the D600 or D6000 master. *For more details consult the manual supplied with the speakers.*

The 601 passes control commands to the current source product. The following commands operate only when one of the CD logos is displayed. *For more details see the section on customising source displays on page 28.*

To start a disc playing

1. Press **Play**

To move on to the next track on a CD

1. Press Play

To pause or unpause a CD

1. Press Pause

To stop a CD playing

1. Press Stop

To select a track directly

Use the number keys on the Remote control as follows:

1. Press one or two number keys
2. Wait one second for the CD player to action the selection

   or

   Press Play to move directly to the selected track.

Controlling the Tuner

If you have a Meridian tuner, you can change the preset station from the 601 remote control.

Note If you have an installation including D600 or D6000 then the speaker takes control from the 601. In these installations the 601 remote control works but the controlling displays are on the D600 or D6000 master. For more details consult the manual supplied with the speakers.

The 601 passes control commands to the current source product. The following commands operate only when Radio is the selected source on the 601. *For more details see the section on customising source displays on page 25.*

To select a preset station

1. Press one or two number keys on the 601 Remote control

   The 601 responds with a select display like:

   | P 1 |

2. Then wait a second for the 601 to action the command

If the first number you enter is 1 or 2, 601 will wait one second for a following number before selecting the preset.

Note Meridian tuners in '204' mode have 18 presets numbered 1 – 6; 11 – 16 and 21 – 26. Other numbers will be ignored.

Operation summary

On the 601 front panel

Key	Action
Off	Puts the 601 into the standby state.
Source	Select or review the sources available.
DSP	Select or review the DSP programmes.
Display	Cycle the display through: source and volume number 2nd volume number track number with index disc elapsed time track elapsed time sampling frequency display off.
Mute	Switches between mute and demute.
Volume Down	Decrease the volume.
Volume Up	Increase the volume.

With the Remote Control

Source keys	Selects input depending on customisation.
Standby	Puts the 601 into the standby state.
DSP	Select or review the DSP programmes.
Display	As on front panel
Phase	Inverts absolute phase.
Blue < and >	Move between DSP menus
Blue ∧ and ∨	Change menu settings
Filter	Turn on or off the DSP actions.
Memory	Recall stored settings
Reset	Restore default settings
Play, Stop, Pause	Control the CD player
Number keys	Numbers sent to the current source
< Left	Moves the balance to the left
> Right	Moves the balance to the right
Red ∧ and ∨	Change the primary volume.
Yellow ∧ and ∨	Change the secondary volume.
Mute	Mutes output.

Customising 601, an overview

601 is a very sophisticated device with a huge number of options to allow you to set up exactly the system you need or to experiment with new possibilities.

Programming 601 is a straightforward process and you should not be timid about experimenting. Although the setup is stored in non-volatile memory called EEPROM, the EPROM (which you cannot change) contains default *Type* settings and you can return to these at any time.

Restoring default settings

1. Switch off any power amplifiers that are connected to 601.
2. Switch off the 601 using the power switch on the back.
3. While pressing the Off key on the 601 front panel, switch the power on again. The display will read in turn:

   Bypass

   (or the name of the last DSP programme); then:

   Type 1

4. Release the Off key.

   At this stage you can change the *Type* from 1 – 6 using the ∧ and ∨ Volume keys; see details on page 65.

5. Switch off the 601 using the power switch on the back.
6. After half a second, switch the 601 power on again.

Caution. Some power amplifiers are very susceptible to small surges that may result from the process of switching a preamplifier on and off. Some – like the Meridian 205 or 605 incorporate DC-offset protection that may be triggered by this process. Meridian active or digital loudspeakers are inherently proof against these surges and need not be switched off. In general digital products like power amplifiers, recorders etc. will be unpredictable when they have the digital signal interrupted.

601 has three setup options; *Type*, *Config* and *Setup*. These together with *Normal* and *Standby* give the 601 five operating states. The following sections give an overview.

Type: an overview

Type just described, restores one of the default settings. No further actions are possible. *Type* always reloads *Bypass*.

The Type settings are described in detail on page 65.

To use Config, see page 25.

For example Type 1 setup gives:

- analogue variable control on the first output
- analogue and digital fixed level on the second output.

Config: an overview

Config mode is used for setting fundamentals of signal types and routing. Consequently in *Config* the outputs of 601 are muted while you make the necessary choices. The choices available in *Config* are outlined in the following sections.

Config: control settings

In *Config* you can make selections that effect the way 601 responds to certain keys. The main choices of this kind are:

- whether you use D600 or D6000 Digital loudspeakers. See page 65.

- whether balance should use delay-compensation. See page 68.
- some display options. See page 27.
- output configurations; fixed, analogue-variable, digital-variable. See page 16.

Details of these are described in the sections referred to.

Config: sources and inputs

In Config you will make selections to do with sources, e.g.

- how many inputs (up to eight) will be active
- which sources you can select
- which key on the remote control selects which source
- the Logo for that input on the 601 display
- the physical input connection to 601 including by implication whether they are analogue or digital.
- the signal and sensitivity of the input

For all the standard *Type* settings, the inputs are reset as follows.

Remote key	Logo	Input	Signal
CD	CD	D1	Digital with display
Radio	Radio	Radio	Analogue, 1V
LP	LP	LP	Analogue – MM, 4mV
Tape 1	TP1	Tape 1	Analogue, 2.0V
Video	Video	Video	Analogue, 2.0V
Tape 2	TP2	Tape2	Analogue, 2.0V

Config: outputs

In this mode you also select the output type for the 1st and 2nd outputs.

601 has both digital and analogue outputs; and both are always available with similar signals for each of the 1st and 2nd output pairs. The output options are an extremely important choice. For each of the outputs – first or second – the following choices are independently available.

- **Analogue variable.** Used to drive analogue power amplifiers or active loudspeakers. Can also be useful for feeding analogue recording machines or headphones.
- **Digital variable.** A variable-level digital signal intended for feeding digital power amplifiers or digital loudspeakers that, unlike D600 or D6000 do not have a built-in volume function. Digital variable is also useful for digital recording since 601 provides a very high-quality fade function in the digital domain.
- **Fixed.** Both analogue and digital outputs operate at a fixed level. This option is used for feeding digital or analogue recording machines, active speakers or digital amplifiers or speakers (like D600 or D6000) that incorporate volume control. For fixed outputs you can select from a wide range of nominal output voltages – these effectively define the gain of the preamplifier.

Setup: an overview

To use Setup, see page 28.

Setup mode is quite different from Type or Config in that the preamplifier actually functions in this mode. In Setup the front panel controls operate normally. The menus are extended to give you additional control.

Output matrix

The first and second outputs can be independently configured for:

- **Stereo** gives normal two-channel behaviour from input to output.
- **Mono** In mono both left and right outputs are the same and are computed from the two input channels as ½(L+ R). This is the correct setting for playing monaural discs.
- **Haffler™** is a simple form of surround system where the rear channels are fed difference signals. The left-rear speaker is fed ½(L–R); the right-rear is fed ½(R–L). With 601 you may choose to feed the rear channels from the secondary output, in which case you would select:

2 Haffler

- **Mono L** In this mono mode only the left input channel feeds both left and right outputs. This mode is sometimes more useful than normal Mono. (Only available in *Archive* currently).

Linked outputs

When 601 has been setup with first and second outputs both variable (digital or analogue), there is a choice to link the volume controls. This is obviously useful for surround installations and some recording applications. When the outputs are linked:

- Volume is controlled by the front-panel ∧ and ∨ keys or by the red ∧ and ∨ keys on the **Remote** control.
- Front-Rear balance is controlled by the **yellow** keys on the **Remote** control.

Filtering

For further details of filtering control see each DSP section following page 38.

In most DSP programmes there is the option to apply the DSP function to the outputs independently. The choices are:

- Filter both
- Filter 1 only
- Filter 2 only

Some common reasons for separating the filtering may be:

- Using the DSP to modify the signal for your speakers, e.g. *Kube™* or *Speaker* DSP programmes, while not doing the same processing on headphones or recorders attached to the other output.
- Using DSP filtering to modify the signal fed to a recorder but not that fed to your speakers.
- Using DSP to modify the headphone output only.

Output level

For more details on analogue output level, see page 29.

Any output selected as 'fixed' is not controlled by the volume keys. Sometimes muting does apply to these outputs –see the section on muting on page 67. When an output is fixed there are no adjustments to be made to the digital version of that output. The analogue output can however be adjusted to suit the following equipment. 601 displays this as a voltage, for example in *Type 1* there is a menu that says:

A2 2.0V

This means that the second analogue output will give 2V rms, for a full-level digital signal. You can adjust this in 15 steps between 450mV and 18V.

For more details on analogue input levels see page 28.

Input level

Digital inputs have, by definition, fixed sensitivity defined by the numbers in the digital data. Analogue inputs however should have their levels adjusted so that the best use is made of the dynamic range of the Analogue-to-Digital converter built in to 601. Best results will be obtained if the analogue gain for each input is set to provide the largest signal that does not cause overloading or clipping.

The 'flat response' analogue inputs like Radio, Video etc, have three choices for sensitivity; 0.5V, 1.0V and 2.0V for full-scale. Phono inputs like Moving Coil or Magnetic (MC or MM) also have three settings.

Note An important feature of *Setup* is that you can adjust the analogue input and output settings while listening to the music. When you feel you have the correct combination of settings you can revert to *Normal* which protects the settings.

Note The input sensitivities for the 601 digital preamplifier are not expressed in the same way as those for analogue counterparts. Caution should therefore be used in comparing these sensitivities. The input level for the 601, in common with other digital equipment is expressed in terms of full-scale in the digital signal – i.e. more equivalent to the input overload point of an analogue preamplifier. See specifications on page 70.

How to customise 601

601 needs no customising to become operational. Once you have your system established you may wish to review the customising options in the section 'Customising 601, an overview', page 18.

Although the 601 comes configured as a standard analogue – analogue preamplifier, there are a very large number of different configurations possible. This section tells you how to change the configuration.

Customising, general procedure

601 has five operating modes (not to be confused with setup *Types*):

- *Normal*
- *Standby*
- *Type*
- *Config*
- *Setup*

Normal and *Standby* are the everyday operating modes: the basic operating instructions refer to these.

Type, *Config* and *Setup* are used to make choices about the way 601 works. These are described in the section: 'Configuring 601, an overview' on page 18.

How to change to *Type*, *Config* or *Setup*

To change to *Type*, *Config* or *Setup* you use the same basic procedure.

1. Switch off any power amplifiers that are connected to 601.
2. Switch off the 601 using the power switch on the back.
3. While pressing the specific key for the operating mode you want on the 601 front panel, switch the power on again.

   601 will display a message to inform you which mode it is in.
4. Release the key.

   At this stage you can make the adjustments required.

In step 3, the front panel key is:

- *Type* – press and hold **Off**.
- *Config* – press and hold **Source**.
- *Setup* – press and hold **DSP**.
- *Normal* – do not use any key!

Caution. Some power amplifiers are very susceptible to small surges that may result from the process of switching a preamplifier on and off. Some – like the Meridian 201 or 605 incorporate DC-offset protection that may be triggered by this process. Meridian active or digital loudspeakers are inherently proof against these surges and need not be switched off. In general digital products like power amplifiers, recorders etc. will be unpredictable when they have the digital signal interrupted.

How to change back to *Normal*

To revert to *Normal* .

1. Switch off the 601 using the power switch on the back.
2. After half a second, switch the 601 power on again.

A full customising process

A full customising process will generally involve using all the configuring modes of 601. A full procedure is as follows.

1. Change to *Type* and select the configuration closest to your requirements.
2. Change to *Config* and make choices about:
   - keys, displays, inputs.
   - output configurations for outputs 1 and 2, fixed, digital variable or analogue variable.
   - control options; D6000?
   - display options; time displays, sampling frequency
   - balance using delay option
3. Change to *Setup* and make choices about:
   - output matrix; Stereo, Mono, Hafler™, Mono L.
   - output linking for volume control
   - filtering applied to first and second outputs
   - analogue output voltage level
   - analogue input sensitivity.
4. Still in *Setup* use the preamplifier.
5. For each DSP programme decide which menu options suit your requirements best.
6. Store these in the memory.
7. Change back to *Normal*.

Making changes to an existing setup

Config and *Setup* are configuration editing modes, they do not reset on entry. *Type* however performs a full reset of all parameters.

To make any changes to your custom configuration:

1. Change to *Config* or *Setup* – whichever is appropriate,
2. make the changes you want,
3. change back to *Normal*.

Customising using Type

For an overview of Type, see page 15.

Type is a mode used specifically to setup 601 into one of six factory-preset configurations. *Type* resets the following for all six *Types*.

- DSP programme to **Bypass**.
- All DSP menus to the default settings. E.g. all tone control features defeated.
- Balance to central.
- Balance with delay option.
- Time displays active in CD sources.
- Sampling frequency not displayed.
- Absolute phase to positive.
- Analogue voltage output level to 2V (for fixed outputs).
- Source Logo input mapping as shown below.
- Depending on the *Type* selected, the outputs are set as follows:

Type	Application	Output 1	Output 2
1	Standard preamp; 2nd fixed output used for recording or Multiroom.	A-variable	Fixed A and D
2	Main speaker with independent headphone or as two-room analogue system.	A-variable	A-variable separate.
3	D600 or D6000, both outputs fixed for recording or Multiroom.	Fixed A and D	Fixed A and D
4	D600 or D6000, 2nd output used for headphone or second-room.	Fixed A and D	A-variable
5	Recording and professional applications.	D-variable	D-variable separate
6	4 speaker system for surround systems or Multiroom.	A-variable	A-variable linked.

Details of muting are given in the table on page 67.

Selecting Types

1. Switch off any power amplifiers that are connected to 601.
2. Switch off the 601 using the power switch on the back.
3. While pressing the **Off** key on the 601 front panel, switch the power on again. The display will read in turn:

   | Bypass |

   (or the name of the last DSP programme); then:

   | Type 1 |

4. Release the Off key
5. Use the front-panel ∧ and ∨ Volume keys to move the selection between *Types* 1 – 6.
6. When you have the *Type* you want displayed, wait one second for 601 to action the choice.
7. Switch off the 601 using the power switch on the back.
8. After half a second, switch the 601 power on again to restore *Normal* mode.

Customising using *Config*

The 601 is initially programmed to *Type 1* and all *Types* set up the source configuration as follows.

For an overview of Config, see page 18

Remote key	Logo	Input	Signal type and level
CD	CD	D1	Digital, subcode display
Radio	Radio	Radio	Analogue, 1V
LP	LP	LP	Analogue – MM, 4mV
Tape 1	TP1	Tape 1	Analogue, 2.5V
Video	Video	Video	Analogue, 2.5V
Tape 2	TP2	Tape2	Analogue, 2.5V

For each source selection key on the Remote control you choose a display *Logo* and a physical input connection – and by implication – signal type. Normally as shown we set the CD key to give a CD display with the D1 (*Digital 1*) input selected. You may wish to display CD1 for your main CD player and to connect it to O1 (*Optical 1*) or one of the analogue inputs.

A full list of possible logos is:

CD	**Radio**	**LP**	**VCR**	**Video**
DAT	**SAT**	**Line**	**TP1**	**TP2**
TP3	**CD1**	**CD2**	**CDR**	**None**

The special **78** logo is only visible in *Archive* with a coarse-groove option selected.

Some of the *Logos* have special meanings, as follows.

- You can only control a *Meridian* CD-player from the 601 Remote control, (see page 15) when you use one of the CD *Logos* i.e. **CD**, **CD1**, **CD2** or **CDR**.
- You can only control a *Meridian* tuner if the source *Logo* is **Radio**.

How to use *Config*

1. Switch off any power amplifiers that are connected to 601.
2. Switch off the 601 using the power switch on the back.
3. While pressing the **Source** key on the 601 front panel, switch the power on again. The display will read in turn:

Bypass

(or the name of the last DSP programme), then:

Config
CD CD

4. Release the **Source** key, the display will show:

Choosing source logos

In Config mode:

1. Press Display until you see a Remote-key (source) + logo display like:

   `CD CD`

2. Press **Source** until the Remote key name for the source you wish to configure is displayed.
3. Use the ∧ and ∨ **Volume** keys until the correct logo is indicated – see the list above.
4. Make changes for all the source options.

Note: If you select **None** the 801 will ignore that key on the Remote control and that source will not be offered in the Normal source menu.

Changing the input for a source

In Config mode:

1. Press **Display**,

you should see the display change to one like:

`CD D1`

The Remote control key you are setting up is the source name and it is indicated on the left as before. Now the letters on the right indicate which physical input connector will be selected. The Display key will toggle between logo and physical input.

A full list of physical inputs is:

D1	Digital 1, coaxial cable	**A–RD**	Analogue Radio
D2	Digital 2, coaxial cable	**A–MM**	LP (magnetic pickup)
D3	Digital 3, coaxial cable	**A–MC**	LP (moving-coil)
D4	Digital 4, coaxial cable	**A–T1**	Analogue Tape 1
O1	Optical 1	**A–T2**	Analogue Tape 2
O2	Optical 2	**A–VI**	Analogue Video

To change the physical inputs:

1. Use the ∧ and ∨ **Volume** keys until the correct one is indicated
2. Make changes for all the source options by pressing **Source**.

Note: When customisation is complete, switch the power off and then on again to restore *Normal* mode and to save what you have done.

Selecting D6000 mode

For more details of D6000 and D600 mode see page 65.

In Config mode:

1. Press **DSP** until you see a display like:

   `D6000? N`

1. Use the ∧ and ∨ **Volume** keys to choose Yes or No

Selecting the type of outputs

In Config mode:

1. Press **DSP** until you see a display like:

   `1 A-var`

This is the setup for output 1.

2. Use the ∧ and ∨ **Volume** keys to choose between:
   - Analogue variable, A. var
   - Digital variable, D. var
   - Fixed
3. Press **DSP** until you see a display like:

   | 2 A.var |

   This is the setup for output 2.
4. Use the ∧ and ∨ **Volume** keys to choose between:
   - Analogue variable
   - Digital variable
   - Fixed

For more details on the implications of these choices see the section on volume controls on page 66.

Note When customisation is complete, switch the power off and then on again to restore *Normal* mode and to save what you have done.

Choosing display options

In Config mode:

1. Press **DSP** until you see a display like:

   | Fs? N |

   This is the choice to have the sampling frequency of digital sources available on the **Display** key.
2. Use the ∧ and ∨ **Volume** keys to choose Yes or No.
3. Press **DSP** until you see a display like:

   | Clip? Y |

   This is the choice to have Clip (overload) messages from the DSP.
4. Use the ∧ and ∨ **Volume** keys to choose Yes or No.
5. Press **DSP** until you see a display like:

   | Time? Y |

   This is the choice to have the time displays for CD sources available on the **Display** key.
6. Use the ∧ and ∨ **Volume** keys to choose Yes or No.

Selecting delay with balance

In Config mode:

1. Press **DSP** until you see a display like:

   | Bdel? Y |

This is the choice to have delay incorporated with balance.

2. Use the ∧ and ∨ **Volume** keys to choose Yes or No.

For more details on this choice see the section on balance on page 68.

Note When customisation is complete, switch the power off and then on again to restore *Normal* mode and to save what you have done.

Customising using Setup

In *Setup* the 601 operates as a preamplifier, there are two major differences from *Normal* mode.

For an overview of Setup see page 19.

- You can adjust, whilst listening, the input and output levels.
- You can store DSP menu settings in permanent memory. These settings are those recalled by the **Memory** key in *Normal*.

How to use *Setup*

1. Switch off any power amplifiers that are connected to 601.
2. Switch off the 601 using the power switch on the back.
3. While pressing the **DSP** key on the 601 front panel, switch the power on again. The display will read in turn:

   `Bypass`

   (or the name of the last DSP programme), then:

   `Setup`

4. Release the **DSP** key, the display will show the normal *Standby* condition:

   `.`

5. Now use the 601 normally until you are happy with the selections described in this section.
6. Store the DSP menu settings you want to keep as 'defaults'.
7. When customisation is complete, switch the power off and then on again to restore *Normal* mode. In *Normal* mode the settings you have made will be retained and cannot be changed without going back to *Setup*.

What does * mean in the menus?

Occasionally in *Setup* you will come across a menu display which has a * character embedded in it, e.g.

`F*both` The * means that for some reason the option is not valid in this DSP programme although it may be available in others. This display does not present a problem, 601 always selects a sensible alternative in these circumstances.

In the example above, * means filtering cannot be applied to both outputs.

Selecting which outputs have DSP

601 has two separate output pairs, first and second. Depending on your installation, you may wish to apply the current DSP function to only one of these outputs.

To select output filtering in *Setup*:

1. Change to the DSP programme you want to use.
2. Use the ← and → blue keys on the Remote control to find the filter configuring menu.
3. Use the ↓ and ↑ blue keys on the Remote control to select between:
   - F both
   - F 1 only
   - F 2 only
4. When customisation is complete, switch the power off and then on again to restore *Normal* mode.

Here are some guidelines.

*For more details about **Headphone**, see page 44.*

- In **Headphone** only apply filtering to the output to which the headphones are connected. If you use the headphone socket on 601 it is connected to the 2nd output. The DSP used for **Headphone** is not at all suitable for loudspeaker listening. In this case you would use:

F 2 only

*For more details about **Kube** see pages 38 and 41.*

- In **Kube or Speaker** where the other output is used for headphones, recording or feeding a second room, apply filtering only to the output feeding the speakers you want to adjust. In this case it will probably be:

F 1 only

*For more details on **Octave Equaliser** see page 55.*

- In **Octave Equaliser** you may be correcting for a room/speaker problem but do not want this correction to be applied to recordings made from the 2nd output. In this case choose:

F 1 only

- In **Octave Equaliser** you may be correcting for a problem in the incoming signal, and want this correction to be applied to recordings made from the 2nd output as well as on the loudspeakers connected to the first output. In this case choose:

F both

*For more details on **Loudness**, see page 60.*

- In **Loudness** it would be usual to apply the DSP filtering to the main loudspeakers – say connected to the 1st output – but not to a fixed output used for feeding another room or a recorder. In this case choose:

F 1 only

Setting the analogue level for fixed outputs

When you have selected an output to be fixed, the level is not adjusted by the volume control.

If you use the digital output there is nothing else to consider because the digital signals are always fed out at the highest level in the digital domain.

In cases where you use the analogue output, the nominal voltage (equivalent to a full-scale digital signal) can be adjusted in Setup to help matching with following components.

To change the analogue output level

In Setup:

1. Use the ∧ and ∨ blue keys on the Remote control to find the analogue output voltage configuring menu. It looks like:

A2 2.0V

   In this case the 2nd output will have a full-scale output of 2Vrms.
2. Use the ∧ and ∨ blue keys on the Remote control to select a level between 450mV and 18V.
3. When customisation is complete, switch the power off and then on again to restore Normal mode.

Note Settings for nominal analogue output above 7.0V must be used with caution. These are provided to give high-gain options, however the 601 analogue output will clip at around 8Vrms so the settings 10.0V, 14.0V and 18.0V will clip on full-scale signals.

Tip Choose the voltage that gives the best overall performance with the following component. A lower output voltage may result in lower noise in the system.

Setting the sensitivity for analogue inputs

All the analogue inputs to 601 are converted to digital signals in a precision Analogue-to-Digital (A/D) converter. The best results will be obtained if the largest possible signal is fed to the A/D converter. To allow this to happen we have provided adjustable analogue input sensitivity.

By setting the sensitivity you also arrange for all sources to be approximately equivalent in volume when you switch between them.

To help you set the inputs we have provided a red light on the front panel labelled 'Level'. The Level light illuminates when the peak input signal is 3dB below (about 70%) full scale.

To change the sensitivity of analogue inputs

For details on how to best adjust the sensitivity of the LP input for turntables, see 'How to get the best out of LP' on page 66.

In *Setup*:

1. Choose the first analogue input you want to adjust, e.g. select **Radio**.
2. Use the < and > blue keys on the Remote control to find the analogue input-voltage configuring menu. It looks like:

   `RAD 1.0V`

   In this case the Radio input will have a sensitivity of *1.0Vrms* for full-scale.
3. Use the ∧ and ∨ blue keys on the Remote control to select levels of 0.5V, 1.0V or 2.5V.
4. Use the 601 to listen to a wide range of material – in this case try a few radio stations. Set the sensitivity to the lowest voltage that you can. A good setting is the lowest one where the Level light flashes rarely or never.
5. Repeat this process for the other analogue inputs.
6. When customisation is complete, switch the power off and then on again to restore *Normal* mode.

Here are some guidelines:

- Meridian tuners; use a sensitivity of 1.0V.
- Inputs from fixed-level digital sources like CD players, D/A converters, digital tuners etc.; use 2.5V.
- Inputs from tape recorders or VCRs; start with 2.5V.

The strongest reason to have the maximum input fed to the A/D converter is to have a good loudness match between analogue and digital sources. The 601 signal path analogue-analogue has a full 18bit dynamic range. It will be most unusual for an analogue signal to have this dynamic range. For example FM radio is equivalent to 13bits and vinyl LPs only achieve the equivalent of 11bits. If you are particularly concerned not to overload the A/D converter then experiment with lower sensitivities (i.e. bigger voltage numbers). See also page 66.

Setting the output matrix: stereo, mono, *Ambio*™, Mono L

For an overview of the output matrix options see page 29.

In *Setup*:

1. Use the < and > blue keys on the Remote control to find the output matrix control menu. In most DSP programmes only the 2nd output is offered; in Headphone this option is not available. It looks like:

   `2 Stereo`

   In this case the 2nd output will be fed a normal stereo signal.
2. Use the ∧ and ∨ blue keys on the Remote control to choose between:

- Stereo
- Mono
- Hafler
- Mono L

3. When customisation is complete, switch the power off and then on again to restore *Normal* mode.

Linking variable outputs

For an overview of the linked output option, see page 20.

This option is only available if you have configured both outputs, 1 and 2, to be variable.

Linking or separating outputs

In Setup:

1. Use the < and > **blue** keys on the Remote control to find the output linking option menu. It looks like:

   `Separate`

2. Use the ∧ and ∨ **blue** keys on the Remote control to toggle between:
   - Separate
   - Linked
3. When customisation is complete, switch the power off and then on again to restore *Normal* mode.

Storing DSP options in memory

Setup mode offers you the chance to store DSP menu settings in permanent memory. Once settings are stored like this they are:

- remembered even when the 601 is switched off from the power for extended periods.
- recalled in *Normal* mode by using the **Memory** key.
- restored automatically whenever you change the DSP programme.

To store DSP settings

For an overview of the memory feature, see page 14.

In Setup:

1. Change to the DSP programme you want to use.
2. Use the < and > **blue** keys on the Remote control to select DSP menus.
3. Use the ∧ and ∨ **blue** keys on the Remote control to select the settings you want.
4. During this process you may find it useful to review the changes you have made by switching the DSP on and off with the **Filter** key.
   When you are satisfied with the settings for that DSP programme:
5. bring up a DSP menu by using the review function of the < and > **blue** keys on the Remote control.
6. While the menu is displayed, press **Memory**. The 601 will display:

   `Store`

7. When customisation is complete, switch the power off and then on again to restore *Normal* mode.

Note You can only store when a DSP menu is displayed. If you press **Memory** at any other time, you will recall the previous setting.

Note When you store a setting, all settings in place for the current DSP programme are stored. It does not matter therefore which menu is displayed when you press **Display**.

Audio connections

There are four types of connection that need to be made to the 601:

- power, see page 8.
- analogue audio, see page 32.
- digital audio, see page 33.
- communications, see page 35.

Analogue audio connections

Analogue connections are made:

- from analogue sources to the analogue inputs of the 601.
- from either of the analogue outputs of 601 to your power amplifiers, active loudspeakers or analogue recording devices like tape-recorders.

Analogue high-level inputs

Analogue inputs should be connected using high-quality screened cable. For advice on the type of cable we suggest you consult your dealer and the manual supplied with the source equipment.

Warning. We strongly recommend the use of well-screened coaxial cable. Certain 'audiophile' cables are not screened. Screened cable will minimise RF interaction with your system and will give more predictable results.

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1. Make analogue connections from each of your sources to an appropriate analogue input on 601. Take care to connect Left and Right correctly. The normal conventions are:
   - left channel is the top one of a pair.
   - right channel is marked red.

For details on adjusting the sensitivity of analogue inputs see page 29.

The high-level 601 analogue inputs are labelled:

- Radio
- Video
- Tape 1
- Tape 2

These inputs are electrically identical and can have their sensitivity adjusted in Setup.

Turntable connection

For details on choosing magnetic or moving-coil options see page 29.

1. Connect the pickup connections to the 601 input labelled LP.
2. Connect the turntable or tone-arm ground wire to the technical ground socket next to the LP input using the 4mm plug included with 601.
3. Configure the input to moving-coil if necessary (using Config as described on page 26).

   601 is supplied set for standard magnetic pickups rated at 4mV for 5cm/s @ 1kHz.
4. Adjust the input sensitivity in Setup as described on page 29.

Analogue outputs, normal

The analogue outputs should be connected using high-quality screened cable. For advice on the type of cable we suggest you consult your dealer and the manual supplied with the associated equipment.

Warning: We strongly recommend the use of well-screened coaxial cable. Certain 'audiophile' cables are not screened. Screened cable will minimise RF interaction with your system and will give more predictable results.

1. Make analogue connections from each output of 601 to the equipment you have selected to follow it. Take care to connect Left and Right correctly. The normal conventions are:
   - left channel is the top one of a pair.
   - right channel is marked red.

The 601 analogue outputs are labelled:

- **ANALOGUE OUT A1**, for output 1.
- **ANALOGUE OUT A2**, for output 2.

These outputs are independent from each other, but carry the same information as the corresponding digital outputs.

Analogue output, balanced

Output **A1** is also available in balanced form on the XLR sockets labelled **L** and **R**. To use the balanced output you will need a balanced cable and a following product that can benefit from this form of connection. For this cable seek your dealer's advice.

Connection to the XLR follow AES and other international standards, namely:
1. Shield and ground or common
2. Signal phase positive
3. Signal phase negative

Meridian products that support balanced connection include:

- M1, M2 and M3 active loudspeakers
- M10, M100, M20 and M30 active loudspeakers
- D600 Digital active loudspeakers
- 605 Power amplifier

Analogue output, headphones

There is a standard 1/4" stereo headphone outlet on the back panel designed for high impedance headphones.

The headphone socket is connected to output **A2**.

Note: This output is for monitoring and is not really intended for serious high-quality headphone listening. For that you should use a separate power amplifier or headphone amplifier. It is our opinion that to fit power amplification for headphones to 601 would compromise its overall sound quality.

Digital audio connections

Digital connections are made:

- from digital sources to the digital inputs of the 601
- from the digital outputs of 601 to your digital power amplifiers, digital loudspeakers or digital recording devices like DAT, DCC or CDR.

Digital connections should be made with either:

- high-quality 75Ω screened cable
- optical fibre to EIAJ standard, e.g. TOSLINK™.

For advice on the type of cable we suggest you consult your dealer and the manual supplied with the source equipment.

Meridian has a range of suitable digital interconnects available as accessories. We recommend you do not use cables intended for analogue connection since these do not usually have the degree of shielding or an impedance of 75Ω. Cable intended for UHF applications like antenna down-lead is also unsuitable since it does not shield adequately in the 1–30MHz region.

Warning. We strongly recommend the use of well-screened precision coaxial cable. Certain 'audiophile' cables are not screened. Screened cable will minimise RF interaction with your system. Unscreened cables for digital connections may cause illegal levels of RF interference.

Digital input connection

For details on assigning the digital inputs to source keys see page 26.

1. Make digital connections from each of your digital sources to an appropriate digital input on 601. The digital inputs on 601 are labelled:

- **D1** to **D4** for the four cable inputs.
- **O1** and **O2** for the two optical inputs.

In the standard setup **D1** is assigned to **CD** and the others are not selected. These inputs are identical in performance and are assigned in Config.

Digital output connections

1. Make digital connections from each output to the equipment you have chosen to follow it.

- **DIGITAL OUT 1** and **DIGITAL OUT 2** for the two cable outputs.
- **DIGITAL OUT 1** and **DIGITAL OUT 2** for the two optical outputs.

These outputs are functionally identical and are assigned in Config.

Note. The optical and cable outputs can be used simultaneously. You therefore have two digital connections to the first output and two to the second.

Communications connections

Communications connections are only made to other Meridian equipment. The following sections describe the major points.

Systems using a Meridian CD player or transport

1. Connect the digital output of the CD player to an appropriate input of the 601.
2. Connect a **B1-Lead between the** 601 (socket marked 600 COMMS) and the CD player (socket marked **200** or **600 COMMS**).

Note A **B1 lead** is different from the **B-lead** that may have been supplied to you. Ask your dealer for the correct type.

Meridian CD and 204 or 604; connection

1. Connect the digital output of the CD player to an appropriate input of the 601.
2. Connect the 204 audio output to the Radio input or connect the digital output of 604 to one of the 601 digital inputs. (In this case you need to configure the digital input as Radio).
3. Connect an E-Lead between:
   - 204 or 604, socket marked **CD** or **207** and
   - the CD player, socket marked **200** or **600 COMMS**.
4. Connect a **Q1-Lead** between:
   - 601, socket marked **600 COMMS** and
   - 204 or 604 socket marked 201/2 or PRE.

Note A **Q1 lead** is different from the **Q-lead** that may have been supplied to you. Ask your dealer for the correct type.

Meridian CD and D600 or D6000; connection

1. Connect the phono plug on the end of the M-lead to the digital output of the CD player.
2. Connect either one of the digital outputs of 601 to a digital input (probably 6H) on the D600 or D6000s using the phono plugs on the **M-Lead** supplied with the speakers.

   or

   For D600 only, connect either of the analogue outputs of 601 to one of the analogue input on the D600s. You could use **A1**, **A2** or the XLR first output using a special cable.
3. Connect the two DIN plugs on one end of the M-lead to:
   - the CD player, socket marked **200** or **600 COMMS** and
   - 601 socket marked **600 COMMS**.
4. See the section on using D600 or D6000 with 601 on page 65.

Meridian CD, 204 or 604 and D600 or D6000; connection

1. Connect the phono plug on the end of the M-lead to the digital output of the CD player.
2. Connect the audio output from the 204 to the Radio input or connect the digital output of 604 to one of the 601 digital inputs. (In this case you need to configure the digital input as Radio).
3. Connect a Q1 lead (see note above) between:
   - 601, socket marked **600 COMMS** and
   - 204 or 604, socket marked **201/2** or **PRE**.
4. Connect the two DIN plugs on one end of the M-lead to:
   - the CD player, socket marked **200** or **600 COMMS** and
   - 204 or 604, socket marked **CD** or **207**.

5. See the section on using D600 or D6000 with 601 on page 65.

Caution The end of the **M-lead** used to connect to other Meridian products has two different plugs. Care should be taken to choose the correct one for the product and not to force any connections.

Optional Optical Communications

Certain Meridian 600-Series products have an alternative communications system based on an 'optical ring'. The products connected with Optical Communications are arranged in such a way that optical fibres are used to connect the **Optical Comms Output** of one product to the **Input** of the next. The ring is closed by returning the **Optical Comms Output** of the last product to the **Optical Comms Input** of the first.

Optical links are available to connect two products. For example Optical Comms can be used to connect 601 to 602. Two illustrations are given below.

601 with 602 CD transport and Optical Comms.

1. Connect an optical communication link between:
   - the 601 sockets marked **600 COMMS IN OUT** and
   - the 602 sockets marked **OPTICAL COMMS IN OUT**

   Connectors marked with **blue** bands to **COMMS IN** on both.
   Connectors marked with **red** bands to **COMMS OUT** on both.

Any other communications connections are made to the **600-COMMS** socket of the 602. Other connections may include:

- a 204 tuner
- D600 or
- D6000 loudspeakers.

Note Making the connections this way avoids ground-loops in the system and maintains the unique advantage 601 has of no galvanic connection between the analogue inputs and outputs.

602, 204 or 604 and D600 or D6000 using Optical Comms.

1. Connect the 602 with OPTICAL COMMS as above.
2. Connect the two DIN plugs on the **M-lead** supplied with the speakers to:
   - 204 (socket marked **201/2**) or 604 (socket marked **PRE**) and
   - 602 (socket marked **600 COMMS**).
3. See the section on using D600 or D6000 with 601 on page 65.

Caution The M-lead has two different plugs at one end and care should be taken to choose the correct one for the product and not to force any connections.

Systems not using a Meridian CD player

204 or 604 only with 601; connection

1. Connect the audio output of a 204 to the Radio input or connect the digital output of a 604 to one of the 601 digital inputs. (In this case you need to configure the digital input as Radio).
2. Connect a DI-lead (see note on page 35) between:
   - the 601 socket marked **600 COMMS** and
   - the 204 (socket marked 205/2) or the 604 (socket marked **FRE**)

D600 or D6000 only with 601; connection

1. Connect either one of the digital outputs of 601 to a digital input (probably D1) on the D600 or D6000s using the phono plugs on the **M-Lead** supplied with the speakers.

   or

   For D600 only, connect either of the analogue outputs of 601 to one of the analogue input on the D600s. You could use **A1**, **A2** or the XLR first output using a special cable.
2. Connect the Din plug on the **M-Lead** supplied with the speakers to the 601 (socket marked **600 COMMS**).
3. See the section on using D600 or D6000 with 601 on page 65.

204 or 604 and D600 or D6000; connection

1. Connect the audio output of the 204 to the *Radio* input or connect the digital output of a 604 to one of the 601 digital inputs. (In this case you need to configure the digital input as Radio).
2. Connect either one of the digital outputs of 601 to a digital input (probably D1) on the D600 or D6000s using the phono plugs on the **M-Lead** supplied with the speakers.

   or

   For D600 only, connect either of the analogue outputs of 601 to one of the analogue input on the D600s. You could use **A1**, **A2** or the XLR first output using a special cable.
3. Connect the socket marked **600 COMMS** on the 601 to one of the DIN plugs on the M-Lead supplied with the speakers.
4. Connect the other DIN plug on the M-lead to the 204 socket marked **CD** or **207** or the **604** socket marked CD.
5. See the section on using D600 or D6000 with 601 on page 65.

Connecting digital recorders; CDR, DAT or DCC

To help you decide which 601 output to use, see 'Making recordings' on page 64.

1. Connect a digital output on the recorder to one of the digital inputs on 601 as described on page 33.
2. Use Config to connect that input to one of the source keys, see page 26.
3. Connect one of the digital outputs of 601 to a digital input on the recorder.

Connecting analogue tape recorders

To help you decide which 601 output to use, see 'Making recordings' on page 64.

1. Connect the analogue output on the recorder to one of the analogue inputs on 601 as described on page 32.
2. Connect one of the analogue outputs of 601 to an analogue input on the recorder.

Note This connection system bypasses (through the ground of the recorder) the galvanic isolation that 601 maintains between inputs and outputs. This removes some of the sonic advantage of 601. The isolation is one of the many factors giving 601 its superb performance. To avoid this problem see page 64.

DSP: *Bypass*

Bypass is the simplest DSP programme supplied with 601. ***Bypass*** is the equivalent of tone-defeat in a conventional preamplifier.

In ***Bypass*** the only functions supplied by the Digital Audio computer are:

- RIAA equalisation for the LP input
- Volume control including dither, balance and muting
- Overload detection

These functions are in fact common to every DSP programme.

In ***Bypass***:

- there are no menu options
- there are no restrictions on the configuration of outputs 1 or 2
- there are no restrictions on balance delay.
- **Mono L** is not available in ***Bypass***.

DSP: *Kube 107*

Kube™ 107 is a digital implementation of a KEF loudspeaker equaliser. The original implementation is as an analogue add-on device. This filtering, although complex in analogue circuitry, can be made completely transparent by using DSP.

Menus

The menus in *Kube™ 107* are:

LF contour		HF contour	
LF	4.0	HF	4.0
LF	0.0	HF	0.0
LF	-6.0	HF	-6.0

In both menus you can adjust the contour level between −6dB and 4dB in 0.5dB steps using the ∧ and ∨ **blue** keys on the Remote control.

Filter key

The **Filter** key on the Remote control switches all the Kube equalisation on and off.

Restrictions

There are no restrictions in ***Kube 107***.

Using *Kube 107*

LF Contour

All loudspeakers interact with the room they are in. This interaction depends on their position within the room. The LF contour function allows you to optimise the bass level in your listening room and to get the balance that suits you best.

To adjust the LF contour:

1. Use the < and > **blue** keys on the Remote control to find the LF menu.
2. Use the ∧ and ∨ **blue** keys on the Remote control to increase or decrease the amount of contour applied. (You can get some idea of what is happening from the graph below).

3. Reduce the number if the bass sounds too 'full'.
4. Increase the number if the bass sounds too 'thin'.

Once you have established the correct setting you should normally be able to leave it there.

Note: Make a note of your preferred setting. If you change **DSP** programmes, the 601 will reset **LF** contour to 0 unless you have stored it in Setup. See page 31. If you have stored it then when you return to ***Kube 107*** or when you press **Memory**, your chosen value will be restored.

HF Contour

HF Contour allows you to adjust the high-frequency balance. To an extent this will compensate for room absorption.

To adjust the HF contour:

1. Use the < and > **blue** keys on the Remote control to find the HF menu.
2. Use the ∧ and ∨ **blue** keys on the Remote control to increase or decrease the amount of contour applied. (You can get some idea of what is happening from the graph below).
3. Reduce the number if the sounds is too 'bright'.
4. Increase the number if the sound is too 'dull'.

Once you have established the correct setting you should normally be able to leave it there.

Note: Make a note of your preferred setting. If you change **DSP** programmes, the 601 will reset HF contour to 0 unless you have stored it in Setup. See page 31. If you have stored it then when you return to ***Kube 107*** or when you press **Memory**, your chosen value will be restored.

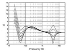
Considerations for *Kube 107*

This programme is designed for equalising loudspeakers to give an improvement in their overall performance. Since it is unlikely that you will have two identical pairs of speakers we recommend that only the output feeding the loudspeakers has filtering implemented when using this programme. To control which outputs have filtering, follow the procedure on page 28.

Although the equalisation in this DSP programme is specifically designed for the KEF 107 loudspeaker, it can be used with other models from other manufacturers. See also ***Kube 200*** on page 41 and ***Speaker equaliser*** on page 57. General points to consider are:

- The loudspeaker must have adequate thermal and mechanical power-handling to withstand the considerable low-frequency boost applied in this programme.
- The power amplifier must have an adequate reserve if you intend to listen at high levels. The low-frequency boost gives a wonderful extension to the loudspeakers, but does so by drawing more power from the amplifier.
- The *Kube* programmes are not generally suitable for active or digital loudspeakers since these usually incorporate low-frequency correction of their own.

Technical data for *Kube 107*



The graph above shows various responses for the *Kube 107* DSP programme. The curve marked with circles is the response you get with HF and LF both set to 0. If the filter is off then the response is exactly 'flat', i.e. at 0dB across the frequency range. The other curves shown are for settings:

LF +4, +2, -2, -4 and -6

HF +4, +2, -2 and -4

DSP: *Kube 200*

Kube*™ *200 is a digital implementation of a *KEF* loudspeaker equaliser. The original implementation is as an analogue add-on device. This filtering, although complex in analogue circuitry, can be made completely transparent by using DSP.

Menus

The menus in *Kube*™ *200* are:

LF contour		HF contour	
LF	6.0	HF	6.0
LF	0.0	HF	0.0
LF	−6.0	HF	−6.0

In both menus you can adjust the contour level between −6dB and 6dB in 0.5dB steps using the ∧ and ∨ blue keys on the Remote control.

Filter key

The *Filter* key on the *Remote control* switches all the *Kube* equalisation on and off.

Restrictions

There are no restrictions in ***Kube 200***.

Using *Kube 200*

LF Contour

All loudspeakers interact with the room they are in. This interaction depends on their position within the room. The LF contour function allows you to optimise the bass level in your listening room and to get the balance that suits you best.

To adjust the LF contour:

1. Use the < and > **blue** keys on the Remote control to find the LF menu.
2. Use the ∧ and ∨ **blue** keys on the Remote control to increase or decrease the amount of contour applied. (You can get some idea of what is happening from the graph below).
3. Reduce the number if the bass sounds too 'full'.
4. Increase the number if the bass sounds too 'thin'.

Once you have established the correct setting you should normally be able to leave it there.

Note: Make a note of your preferred setting. If you change **DSP** programmes, the 601 will reset LF contour to 0 unless you have stored it in Setup. See page 31. If you have stored it then when you return to ***Kube 200*** or when you press **Memory**, your chosen value will be restored.

HF Contour

HF Contour allows you to adjust the high-frequency balance. To an extent this will compensate for room absorption.

To adjust the HF contour:

1. Use the < and > blue keys on the Remote control to find the HF menu.

2. Use the + and – blue keys on the Remote control to increase or decrease the amount of contour applied. (You can get some idea of what is happening from the graph below).
3. Reduce the number if the sounds is too 'bright'.
4. Increase the number if the sound is too 'dull'.

Once you have established the correct setting you should normally be able to leave it there.

Note Make a note of your preferred setting. If you change **DSP** programmes, the 601 will reset HF contour to 0 unless you have stored it in Setup. See page 31. If you have stored it then when you return to ***Kube 200*** or when you press **Memory**, your chosen value will be restored.

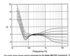
Considerations for *Kube 200*

This programme is designed for equalising loudspeakers to give an improvement in their overall performance. Since it is unlikely that you will have two identical pairs of speakers we recommend that only the output feeding the loudspeakers has filtering implemented when using this programme. To control which outputs have filtering, follow the procedure on page 28.

The equalisation in this DSP programme is matched to several KEF loudspeakers. It can also be used with other models from other manufacturers. See also ***Kube NF***, page 38 and ***Speaker Equaliser*** on page 37. General points to consider are:

- The loudspeaker must have adequate thermal and mechanical power-handling to withstand the considerable low-frequency boost applied in this programme.
- The power amplifier must have an adequate reserve if you intend to listen at high levels. The low-frequency boost gives a wonderful extension to the loudspeakers, but does so by drawing more power from the amplifier.
- The ***Kube*** programmes are not generally suitable for active or digital loudspeakers since these usually incorporate low-frequency correction of their own.

Technical data for *Kube 200*



The graph above shows various responses for the ***Kube 200*** DSP programme. The curve marked with circles is the response you get with HF and LF both set to 0. If the filter is off then the response is exactly 'flat', i.e. at 0dB across the frequency range. The other curves shown are for settings:

LF +4, +2, -2, -4 and -6

HF +4, +2, -2, -4 and -6

DSP: *Headphone*

As the name implies, ***Headphone*** is a programme designed specifically for listening through headphones to material intended for reproduction over stereo loudspeakers.

The basis of headphone mode is explained as follows:

- When we listen in a room to a pair of loudspeakers, both ears pick up sounds from both loudspeakers. When we listen using headphones the left and right signals are fed directly to each ear.
- Sounds coming from the left loudspeaker are different at each ear. The left ear receives the sound first (assuming we are facing the speakers). The right ear receives a later sound and one which is modified by the diffraction or 'shadow' effect of the head. This modification is quite complex and is very dependent on frequency.
- In normal listening the sounds we hear incorporate modifications made by the acoustic properties of our outer-ears or pinnae. The modifications include multiple reflections which combine to give us an acute sense of horizontal and vertical direction.
- In headphone listening the pinnae are nearly always covered up, certain headphones e.g. in-ear devices bypass the pinnae altogether.
- The lack of normal acoustic crosstalk and of pinna effect often lead to unnatural sound quality on headphones.
- There are in addition two major groups of headphones with regard to frequency balance. These are called '*diffuse*' and '*free-field*'. The difference is illustrated later in this section.

The ***Headphone*** programme in the 601 attempts to correct these problems with digital signal processing.

The ***Headphone*** programme offers four standard headphone equalisations, called Types A-D. In addition there is a full Custom mode in which you can experiment to get the best result for your ears and headphones. Custom mode is important since everyone's pinnae are different and these differences are vital to us perceiving sound 'out of head'.

For more details on Custom mode, see the later section 'Technical data for Headphone'.

Menus

Headphone uses a new type of menu structure that allows nested menus. When you first select **Headphone** the choices will be between an overall tone control with a range of ±10dB and the 'Types' menu.

1. Move between Tone and Types with the **blue** < and > keys on the Remote control.
2. Select the tone settings in the Tone menu with the **blue** ^ and v keys on the Remote control.
3. Choose Types in the Type menu with the **blue** ^ and v keys on the Remote control.

In the 'Types' menu there is an entry called 'Custom'. After choosing Custom:

1. Choose between the equalisation and delay controls with the **blue** < and > keys on the Remote control.
2. Select choices in these menus with the **blue** ^ and v keys on the Remote control.

The menu structure in **Headphone** is as follows.

Tone	Type	Forward equalisation					Cross-feed			Scale
		2kHz	3kHz	4kHz	7kHz	10kHz	DTL	DTA	DTh	SCL
12	Custom	12	12	12	12	12	9	12	12	0.0
	A									
	B									
	C									
	D									
0		0	0	0	0	0	0	0	0	
−10		−12	−12	−12	−12	−12	−12	−12	−12	−6.0

Filter key

The **Filter** key on the Remote control switches all the ***Headphone*** equalisation on and off. Use it to compare corrected and uncorrected signals.

Restrictions

Haffer™ and Mono are not available in *Headphone*.

Suggestions for *Headphone*

This programme is designed for equalising headphones to give an improvement in their overall performance. Since it is unlikely that you will be using two separate pairs of headphones we recommend that only the output feeding them has filtering implemented when using this programme. To control which outputs have filtering, follow the procedure on page 28.

Individuals differ a great deal in the kind of adjustments they make in this mode. An incorrect equalisation for you could be worse than none. Take time to choose the settings that suit you best, then store them in permanent memory as described on page 31.

Be aware that the kind of filtering that corrects headphone listening sounds very strange over loudspeakers.

Note Take extra care not to accidentally use ***Headphone*** when making recordings unless the recordings are intended for use with a personal cassette recorder like a Walkman™!

Technical data for *Headphone*

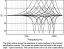
The signal processing is split into three sections:

- **Forward.** Equalisation and correction of the overall signal to both ears from the left and right inputs.
- **Cross-feed.** Equalisation and correction of a cross-feed signal that feeds some of the left input to the right ear and vice versa.
- **Scale** Abbreviated to SCL in the menus. This parameter adjusts the overall relative gain of the ***Headphone*** processing so that you get a useful comparison when switching the processing on and off with the **Filter** key.

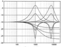
The forward equalisation consists of a special 'graphic' equaliser with filters at 2kHz, 3kHz, 4kHz, 7kHz and 10kHz. These bandpass filters all have an adjustment range of ±12dB in 0.5dB steps.

The cross-feed sections have the following adjustments:

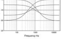
- **Delay.** The signal to the far-ear can be delayed. The delay number can be adjusted between 1 and 63. Each unit represents one sampling period in the digital domain, so e.g. at 44.1kHz each unit is 22.67µs. The initial setting of 13 units give an inter-aural delay of about 300µs which corresponds to a source 30° to one side of dead-ahead. (This is a common speaker position viewed from the listening seat) More delay would bring you closer (in this respect) to the sound-stage.
- **Equalisation.** The cross-feed signal has additional equalisation in the form of two bandpass filters centred on 1kHz and 7kHz. These filters have adjustment ranges of ±12dB in 0.5dB steps. Both filters will adjust individual differences and help to account for image height and 'forwardness'. A common problem in headphone listening is that the image is either perceived inside the head or just above it!
- **Cross-feed tilt.** The cross-feed signal has an overall low-pass characteristic applied to it to simulate the shadow effect of the head. This tilt has the range 0dB to −12dB in steps of 0.5dB.



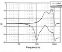
The graph above shows the responses of various settings of the forward equalisation controls. The curves are plotted from left input to left output (right – right is identical). The curves shown are for ±6 and ±12dB settings on each of the 2kHz, 3kHz, 4kHz, 7kHz and 10kHz bandpass filters.



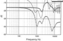
The graph above shows the responses of various settings of the cross-feed equalisation controls. The curves are plotted from left input to right output (right – left is identical). The curves shown are for all and ±12dB settings on each of the 1kHz and 7kHz bandpass filters and for –4, –8 and –12 on DTL – the cross-feed tilt.



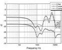
The graph above shows the overall Tone control behaviour in **Headphone**.



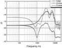
The graph above shows the frequency responses for the forward and cross-feed channels in the Type A standard setting.



The graph above shows the frequency responses for the forward and cross-feed channels in the Type B standard setting. Also shown is the composite result to one output when the left and right input signals are identical. The 'ragged' look to the response at high frequencies is the result of comb-filtering due to the inter-aural delay.



The graph above shows the frequency responses for the forward and cross-feed channels in the Type C standard setting. Also shown is the composite result to one output when the left and right input signals are identical. The 'ragged' look to the response at high frequencies is the result of comb-filtering due to the inter-aural delay.



The graph above shows the frequency responses for the forward and cross-feed channels in the Type D standard setting. Also shown is the composite result to one output when left and right signals are identical.

DSP: *Archive*

Archive is a unique programme for extracting the very best performance from many types of gramophone (phonograph) records. The options offered are:

- Accurate replay for four fine-groove standards including RIAA
- Accurate replay for nine coarse-groove standards
- Correction to these standards for any previous transcriptions made using RIAA. This option allows you to correct any tapes or transfers you may have which were taken using the wrong characteristic.
- An overall tilt-type tone control
- Switchable rumble filters using Bessell characteristics
- A scratch filter.
- A notch filter at 4kHz. This filter is placed at this frequency to counteract the very common cutter-resonance problem seen on many 78s.

Menus

The menus in *Archive* are:

Tone control dB	Replay standard	4kHz notch depth dB	Low-pass 'scratch' filter	High-pass 'rumble' filter
Tone 10	NAB	Notch 0	Scratch	HP 50Hz
Tone 8	AES	Notch –1	No Scra	HP 30Hz
Tone 6	STD 78	Notch –2		HP Flat
	BSI 78	Notch –3		
Tone 2	Brunswick	Notch –4		
Tone 1	Westrex	Notch –5		
Tone 0	FFRR 78	Notch –6		
Tone –1	Decca 78	Notch –7		
Tone –2	Columbia	Notch –8		
	NARTB	Notch –9		
Tone –6	EMI LP	Notch –10		
Tone –8	FFRR LP	Notch –11		
Tone –10	RIAA LP	Notch –12		

Filter key

The **Filter** key on the Remote control does not function in ***Archive***.

Display

When you are using the turntable input (labelled LP) on the back of the 601, the source logo normally displayed is LP. In this circumstance selecting a coarse-groove equalisation automatically changes the source logo to:

78 44	Where the 78 means 78rpm and in this case 44 is the volume number.

Restrictions

In *Archive*:

- there are no restrictions on the configuration of outputs 1 or 2

- there are no restrictions on balance delay.

Suggested uses for *Archive*.

- Replaying and/or recording transcriptions of many types of gramophone record with optimal sound quality adjustable by you.
- Making corrections to previous transcriptions which were incorrectly made using RIAA replay. In this application the signal will be coming in on either a high-level analogue input e.g. from tape, or as a digital signal e.g. from a CD transfer.

Note In *Archive* the filtering is applied to both outputs or not at all.

Technical data for *Archive*

Replay standards

The replay standards are described by the time-constants used in generating the replay characteristic. They are given in the following table.

Replay standard		t_1 µs	t_2 µs	t_3 µs	t_4 µs
NAR	78	7958	2240	318	100
AES	78	7958	n/a	400	64
STD	78	7958	n/a	636	n/a
BSI	78	7958	3180	451	50
Blumlein	78	7958	3180	636	n/a
Western	78	7958	n/a	796	n/a
FFRR	78	7958	3980	636	25
Decca	78	7958	n/a	1063	50
Columbia	LP	7958	1590	318	100
NARTB	LP	7958	n/a	318	100
EMI	LP	7958	2271	318	64
FFRR	LP	7958	1590	318	50
RIAA-IEC	LP	7958	3180	318	75

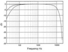
Rumble filter

The rumble filter has a high-pass 2nd-order Bessell response. The roll-over frequency can be set to 30Hz or 50Hz.

Note we elected to leave the IEC 20Hz first-order roll-off in place for all standards since there is no useful information below 20Hz on these recordings.

Scratch filter

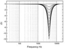
The scratch filter is a 2nd-order low-pass Bessel filter at 10kHz.

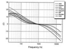


The graph above shows the responses of the scratch and rumble filters.

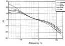
Notch filter

The notch filter is approximately an octave wide and centred on 4kHz.
The notch depth can be adjusted between 0dB (off) and −12dB.

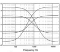




The graph above shows the frequency responses of the coarse-groove 78 rpm equalisations offered in *Archive*.



The graph above shows the frequency responses of the fine-groove 33 rpm equalisations offered in *Archive*.



The graph shows the frequency response of the representative Tone settings ±5 and ±10dB in *Archive*. The same tone responses are fitted to *Speaker Eq*.

DSP: *Octave Equaliser*

Octave Eq. is a digital implementation of an octave graphic equaliser.

Menus

The menus in **Octave Eq.** are:

31 Hz	63 Hz	125 Hz	250 Hz	500 Hz	1k Hz	2k Hz	4k Hz	8k Hz	16k Hz

In each menu you can adjust the response by ±7dB in 1dB steps.

A typical menu display is:

4kHz 4	This would mean that the octave centred on 4kHz has been raised by 4dB.

Filter key

The **Filter** key on the Remote control switches the octave filters on and off.

Restrictions

In ***Octave Eq.***:

- there are no restrictions on the configuration of outputs 1 or 2
- there are no restrictions on balance delay.

Suggested uses for *Octave Eq.*

For details on how to control which outputs are filtered see page 25.

- Correcting for loudspeaker or room response. In this application you may choose to set up the 601 to apply filtering only to the output feeding your loudspeakers. This may be important if you use the other output for making recordings or for headphone listening.
- Making corrections to incoming signals to overcome recording or transmission defects. In this case you would normally set up the 601 to filter both outputs.

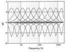
Technical data for *Octave Eq.*

The equaliser in ***Octave Equaliser*** uses a 'side-chain' structure. In a side chain, a filter is set up in a parallel path to the original signal. Varying amounts of the filtered signal are then added or subtracted from the original. There are several advantages to this approach.

- When the controls are set 'flat' i.e. to 0dB, the signal is not passed through a filter and so remains fully intact and has no phase errors.
- The response shapes that result from adding or subtracting a correction are not symmetrical, but bear a more accurate resemblance to real physical circumstances where resonances cause more or less signal to be heard.
- The user can obtain more subtle and natural control through the choice of boost or cut than with a more conventional octave equaliser.
- This approach which is based on correcting for real physical phenomena like room or speaker resonances is deliberately not linear-phase.

Frequency Response of *Octave Equaliser*

The frequency response capability of the 601 ***Octave Eq.*** is illustrated in the following graph. In each case the octaves were set to + or – 7dB separately.



DSP: *Speaker Equaliser*

Speaker Equaliser is a programme designed specifically to help you integrate your loudspeaker to the listening room and to get the most natural sound from it. ***Speaker Eq.*** offers the following adjustments:

- **Tone** A tilt-type of tone control that can give a broadly rising or falling frequency response to the system. For example if the loudspeaker is generally 'thin' or 'bright' in sound – or if the room is very reflective and therefore bright-sounding – this control can help to correct these tendencies. Tone is adjustable in 1dB steps over the range ±10dB using the **blue** ∧ and ∨ keys on the Remote control.
- **Bass** The bass control is not at all like a conventional bass control. The purpose of this is, somewhat like in the Kube™ programmes, to give the speakers a lower bass extension without additionally boosting the upper-bass or lower-midrange regions. The bass control has the characteristics shown in the graphs below; the range covers ±15dB in 1dB steps.
- **Wall** Many loudspeakers are designed for free-space mounting, others are equalised for a nominal position with respect to the near boundaries. Placing a speaker near a wall – which is inevitably necessary – changes its subjective frequency balance because the near boundaries give added acoustic gain at the lower frequencies without affecting the mid or top.

  The more walls near to the speaker the worse it is; and in this context the floor behaves like a wall.

  The frequency range affected depends on the size of the speaker and the distance from the near boundaries.

  Although walls give gain, we have allowed this control to also account for responses where a speaker intended for wall-mounting is in fact away from the wall – and which therefore requires some boost in the signal. The wall control has a range of ±15dB in 1dB steps.

Menus

The menus in ***Speaker Equaliser*** are:

Tone	Wall	Bass
Tone 10.0	Wall 15.0	Bass 15.0
Tone 0.0	Wall 0.0	Bass 0.0
Tone −10.0	Wall −15.0	Bass −15.0

Move between menus with the ‹ and › **blue** keys on the Remote control.

In all menus you can adjust in 1dB steps using the ∧ and ∨ **blue** keys on the Remote control.

Filter key

The ***Filter*** key on the Remote control switches all the ***Speaker Equalisation*** on and off.

Restrictions

There are no restrictions in ***Speaker Eq.***

Considerations for *Speaker Equaliser*

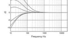
This programme is designed for equalising loudspeakers to give an improvement in their overall performance. Since it is unlikely that you will have two identical pairs of speakers we recommend that only the output feeding the loudspeakers has filtering implemented when using this programme. To control which outputs have filtering, follow the procedure on page 28.

The equalisation in this DSP programme is chosen to cover a very wide range of loudspeakers. See also **Kube 107**, page 38 and **Kube 208** on page 41. General points to consider are:

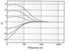
- The loudspeaker must have adequate thermal and mechanical power-handling to withstand the considerable low-frequency boost applied in this programme.
- The power amplifier must have an adequate reserve if you intend to listen at high levels. The low-frequency boost gives a wonderful extension to the loudspeakers, but does so by drawing more power from the amplifier.

Technical data for *Speaker Equaliser*

The frequency response of the bass control in ***Speaker Equaliser*** is illustrated by the examples in the graph below. The illustrations are for settings of ±4, ±8 and ±12dB.



The frequency response of the wall control in ***Speaker Equaliser*** is illustrated by the examples in the graph below. The illustrations are for settings of ±5, ±10 and ±15dB.



The Tone control in ***Speaker Equaliser*** is the same as the one fitted to ***Archive***. The response of that control is illustrated on page 54.

DSP: *Loudness*

Loudness is a DSP programme which adjusts the frequency response of the preamplifier according to the volume level you are using.

The term loudness has a specific meaning in acoustics – or more exactly in the area of study known as psychoacoustics. The loudness of a sound refers only to its magnitude and changes on a scale from soft to loud. Loudness is measured in *sones* or *phons*.

The word 'volume', which, since the 1920s has come to describe the control on virtually all audio devices, has a subtly different meaning. The volume of a sound is a combination impression that includes its loudness and its apparent frequency width or range.

The reason for the distinction in the language is that human hearing has a range that depends very much on the sound energy or intensity. In the technical data section below there is a graph showing the normal threshold for hearing and equal-loudness curves for human hearing. You can see that at higher levels there is a much broader frequency width possible than when the sounds are quiet. You will also see that at the extremes of human hearing e.g. below 40Hz or above 15kHz considerable energy is needed before the sound is audible.

It is felt by many experienced listeners that music reproduction is most natural when the sound is actually reproduced at the original volume, that is, the sound should be as loud as at the original performance.

For many types of music this is either not possible from the loudspeakers – or may be anti-social! A correctly-designed loudness control can deal with this problem. The graph below also shows the 'equal loudness contours' for human hearing. At normal music levels (90 – 110dB spl) the frequency response of human hearing is much flatter than at lower levels.

The *Loudness* function in 601 actually corrects for the difference between the equal-loudness contours when we listen to music at lower levels. This complex filtering essentially boosts the low and high-frequencies progressively as the volume is reduced.

Setting up Loudness

In order to use **Loudness** you have to tell 601 what the nominal volume number is; that is the number at which you feel your power-amplifier + speaker combination gives 'natural loudness'.

To set up the loudness '*threshold*' either calculate it from first-principles as described in the section *Technical data for loudness*, or:

1. Go to Setup mode as described on page 28.
2. Make sure **Loudness** is the current mode.
3. Switch the filtering off using the **Filter** key on the Remote control.
4. Play a CD and adjust the volume to get a natural volume. Remember this will be quite loud, too loud for many people's expectation of domestic music listening. Try using a piano recording or some material with which you have a recent live experience. Make a note the volume number; this will be your threshold setting.
5. Using the < and > **blue** keys on the Remote control, look for the threshold menu; it has a display like:

Thres 78

6. Using the ^ and v **blue** keys on the Remote control, set the number to your chosen threshold.

While you have the threshold display.

7. Press **Memory** on the Remote control to store the number in permanent memory.
8. Go back to *Normal* from *Setup*: switch the 601 off and on again using the power switch on the back panel.

The 601 is now setup for loudness in your installation.

Note In *Digital* variable mode (as opposed to the *Analogue* variable setting in *Config*) the volume numbers indicate between +6dB and -93dB. The threshold is set relative to this range and starts at -29dB.

This programme is designed for equalising loudspeaker listening. We recommend that only the output feeding the speakers has filtering implemented when using this programme. To control which outputs have filtering, follow the procedure on page 29.

Note Take extra care not to accidentally use **Loudness** on an output while making recordings!

Menus

The menus in *Loudness* are:

Loudness	Tone
Loudness	Tone 10
Normal	
	Tone 0
	Tone -10

The Tone control is a tilt-type that can give a broadly rising or falling frequency response to the system. Tone is adjustable in 1dB steps over the range ±10dB using the **blue** ^ and v keys on the Remote control. This Tone control is similar to the one fitted in *AirAlert*, frequency responses are shown on page 64.

Filter key

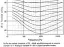
Use the **Filter** key on the Remote control to switch all the DSP on and off.

Note *Filter* off defeats *Tone* whereas Loudness off (displayed as *Normal*) leaves *Tone* settings in place.

Technical data for *Loudness*

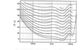
In *Loudness* the 801 computes a frequency response which emulates the difference between the equal-loudness contours of human hearing taking 90phon as the reference. The graph below shows this behaviour, as the volume is turned down below the *Threshold* setting, more and more low and high-frequency boost is applied to the signal.

In the graph above, the dB axis is relative to the *Threshold* you have set.



So for the default threshold of 70, -60dB would correspond to volume number 10 in *Analogue variable* or -60 in *Digital variable* modes.

To have more *Loudness* effect, choose a higher threshold level.



The graph above shows the equal-loudness contours for human hearing. Note how our sensitivity to low frequencies increases more with level than in the mid-range.

The *Loudness* mode of 601 attempts to fit to the difference between these curves and the 90 phon curve - i.e. the curve that passes through 90dB spl at 1kHz.

Making recordings

General

The 601 is ideally suited to making many types of recordings. The particular advantages are listed below.

Advantages for all recordings

- Both analogue and digital recorders can be used at the same time.
- The full power of any of the DSP programmes can be used to adjust the signal fed to the recorder. See following page 38.
- By correctly selecting which outputs have DSP filtering applied you can have DSP on:
  - the speakers only
  - the recording only
  - both speakers and recording

For further details see page 28.

Advantages for analogue recordings

- The analogue output used for recording can be adjusted to a voltage level that suits the recorder best. See page 29.
- The output used for recording can be at a different level from that feeding the speakers by choosing to use the 1st and 2nd outputs correctly. See page 16.

Advantages for digital recordings

- Up to four digital recorders can be fed simultaneously by using the optical and cable outputs.
- The 601 digital volume can be used to create fades in the digital domain. These fades are correctly dithered and present a huge advantage to making digital recordings.
- The 601 offers up to +6dB gain in the digital domain, making it possible for signal levels to be matched in digital recordings.

Tape monitoring

The 601 is unusual in not offering a 'tape-monitoring' facility. The reason for this is that 601 only has one audio DSP computer fitted. Two DSPs would make for considerable increase in cost and give very little benefit. If this facility is essential to you external switching will be needed.

Ground-loops in analogue recording

As pointed out on page 37, an analogue recorder is normally connected to both:

- an analogue input of 601 and
- an analogue output.

This connection places – through the ground of the recorder – a connection between the input and output grounds of 601.

Note This connection system bypasses (through the ground of the recorder) the galvanic isolation that 601 maintains between inputs and outputs. This places 601 at a slight sonic disadvantage. The isolation is one of the many factors giving 601 its superb performance.

To avoid this problem you have two choices:

- Disconnect the input cable to the recorder when you are not actually making a recording.
- Use an outboard D/A converter like the Meridian 203 or 263u2.

To use an outboard converter like 233 or 263(L):

1. Make a digital connection between one of the digital outputs of the 601 and the digital input to the converter.
2. Connect the converter's analogue output to the input of the recorder.
3. Connect the analogue output on the recorder to one of the analogue inputs on 601 as described on page 33.

General caution for making recordings

As you will know by now, 601 is a complex and powerful preamplifier. Before making any recording check the following.

1. Which DSP programme is loaded? If you want to be sure of no processing load **Bypass**.
2. Is the DSP you want on the recording being applied to the output feeding the recorder?
3. Can you adjust the volume or mute the loudspeakers without changing the recording level in the configuration you have chosen?

Tip: While a new DSP programme is being loaded 601 mutes the sound for about one second. This is necessary to allow the computer to synchronise to the data. It means however that you should never change the DSP function while making any kind of recording since there will be a break in the sound.

Using D600 or D6000 speakers with 601

When you use D600 or D6000 speakers with 601 start off with one of the appropriate Type settings – see page 65.

Type	Application	Output 1	Output 2
3	D600 or D6000, both outputs fixed for recording or Multiroom.	Fixed A and D	Fixed A and D
4	D600 or D6000, 2nd output used for headphone or second room.	Fixed A and D	A-variable

In Config you can independently make the choice to have 601 in the D600 or D6000 mode. Setting Type 3 or 4 does this automatically.

Normally you connect the digital speakers to the fixed output of 601; in these types this means use output 1.

In D600 or D6000 mode the following differences apply to 601.

- You can control the complete system with either the Remote control supplied with 601 or with the **609** or **D6000 Remote** supplied with the speakers.
- Volume control and mute are handled by the speakers.
- **Phase** and **Display** depend on which remote control you use.
- Sources can be selected remotely via the loudspeakers or from the front panel of 601.
- Muting can be controlled from the front panel of 601.

Additional technical information

Why do I want two outputs?

The 601 has two separate output channels. This has been arranged to give you the maximum flexibility. A few of the reasons are listed below.

- You can set up surround systems using the two output pairs to give four channels operating together. (See pages 31 and 20).
- You can set up a three-channel system by configuring the second output to be Mono and Linked. Use this signal to feed a centre speaker. In this case you adjust the relative level of the centre speaker with the front-rear balance feature of the yellow buttons on the Remote control. See page 20.
- You can provide two independently variable outputs for driving two rooms or for driving loudspeakers and headphones separately. (See page 31). In addition one of these can be Analogue variable while the other is Digital variable.
- You can provide one variable output while the other can be fixed for feeding a Multiroom zone or recording machine.
- You can have DSP applied to just one of the outputs using the Filter control feature described on page 28. There are several examples of this making sense depending on the DSP programme you are using.
- By setting the outputs to both variable and linked, you can use the two outputs of 601 to drive twin power amplifiers in a bi-amped system. In this case you would normally set the front-rear balance (using the **yellow** keys on the Remote control) to 0 or a value close to 0.

What is the difference between analogue and digital variable?

If you use the analogue output of 601 with 'Analogue variable' having been selected in Config – see page 23 – then the volume will change smoothly over the whole range of volume numbers from 0 to 99. While this is happening however the corresponding digital output will not behave the same way. Between volume numbers 0 and 70, the digital output will increase smoothly but by 70 it will have achieved full-scale. Between 70 and 99 the digital output will go up and down by up to 4dB. The reason for this is that the analogue variable output uses some analogue gain switching to get the maximum resolution and signal/noise ratio on that output.

The best way to explain this is that a 16-bit digital signal has a dynamic range of 96dB and to cover the complete range of human hearing (for music signals) a range of 18.5 bits is required. By using additional analogue switching we ensure that the 601 makes the best fit between precision and inaudible background noise.

If you select 'Digital variable', then both the analogue and digital outputs behave in the same way – that is they both increase smoothly in level from volume number –93 up to +6. In this circumstance the analogue output level is determined by the level set in Setup – see page 29. The main difference now is that the digital output is smoothly controlled over the whole range, this is the intended function. The corresponding analogue output does not give the best precision. For example at volume 70 with 'Analogue variable' the analogue output has full 18 bit precision. In 'Digital variable' the analogue output has about 11 bits precision at the corresponding setting of –23. If you have selected 'Digital variable' for one of the outputs of 601 then the volume displays are different. For example at start-up instead of seeing a volume number of 70, we get:

CD	0

Here 0 is a gain setting in dB. 0dB corresponds to a factor of unity, i.e. the 601 has a one-to-one ratio between the digital input and digital output. If you change the volume the display will show volume numbers in the range –93 to +6. +6dB is a factor of 2 increase, –20dB is a factor of 0.1 and so on.

In 'Fixed' output both the analogue and digital outputs run at full precision.

Understanding input sensitivity in a digital preamplifier

There is no variation of sensitivity for digital inputs. The consumer SPDIF interface defines full-scale in terms of the digital word coming in. Occasionally you may want to increase the level of a digital signal – this can be done in 'Digital variable' mode – see the previous section.

In a conventional analogue preamplifier the input sensitivity is normally quoted as that level required to give rated output when the volume (i.e. gain) is turned full up. In 601 this figure is less important than the maximum input that can be fed to the preamplifier without distortion. This level is sometimes given for analogue preamps as the overload level.

For the analogue inputs of 601, the input sensitivity you select in Setup (see page 29) is the voltage at which the analogue-digital converter will overload. For some sources – e.g. Radio, Tape or Video this can be very closely defined and the nominal output of the source will be within 3dB of its maximum. For other sources – e.g. turntables, the maximum signal is less well defined and some experimentation may be beneficial. See the section following on 'How to get the best out of LP'.

Clipping and overload

There are two types of overload that can occur in 601, at the input and inside the DSP.

Input overload occurs for analogue signals that overdrive the analogue-digital converter. We have fitted a **Level** light to the front panel of 601. If you are close to overload on analogue sources this will flash. The threshold for this light is 3dB below clipping. Ideally you will use a combination of 601 input sensitivity and source output that is as much as possible consistent with this light never – or only very rarely – flashing. An obvious exception to the 'never' rule would be on interference signals like ignition noise on a tuner or bad scratches on a record.

DSP overload can occur if the combination of input signal and the filtering you have requested puts the signal outside the 18 bit dynamic range. We have taken care in the design of the DSP programmes to make this a very rare event. If the DSP reaches overload it performs an elegant and barely-audible clip and the 601 display will momentarily show:

`Clip 1`

Here the 1 refers to output 1. The message may refer to output 2 or to both with:

`Clip 1&2`

To avoid clipping reduce the volume or digital gain slightly.

How to get the best out of LP

The turntable source is quite unique in the problems it presents to getting optimum sound quality. Obviously the turntable itself must be correctly setup and we recommend you have your dealer check it from time-to-time.

601 has several adjustments that affect the sound quality of turntable sources. Here is a starting checklist.

1. Is the 601 installed far enough away from components which contain mains transformers to ensure that you get no hum induced?
2. Is your cartridge Moving-coil or Moving magnet. Remember to set this choice in Config. See page 25.

Pickup cartridges tend to have their output specified at 5cm/s recorded velocity. Normally the maximum velocity found on an LP is 25cm/sec at the outside and around 15cm/s in the inner grooves. 25cm/sec is about the maximum tracking ability of audiophile MC cartridges. Certain moving-magnet cartridges can track up to 35cm/sec but there are rarely these velocities to be found.

On the face of it we could adjust the LP input sensitivity to 5 times the rated output of the cartridge. If we do this then as expected the overload **Level** light very rarely illuminates. This kind of setting gives three benefits:

- The maximum digital level for any recording
- The lowest noise contribution from the analogue-digital converter.
- The best loudness match when switching from LP to other sources.

However, under many circumstances you may prefer to set 601 to a lower sensitivity – i.e. larger number in *Setup* – because you prefer the sound overall. The reason is that the analogue-digital converter has such a wide range compared to the surface-noise of LPs, that you can allow more headroom (for very short clicks due to dust etc.) without any noise penalty. This alternative fitting of the available dynamic-range of the LP signal into the 601 input 'window', can give better sound overall but at the expense of level matching between sources.

Tip: For moving-coil cartridges start at the 600µV setting. For moving magnet use 4mV.

How mute works in different configurations

The table below describes how muting is controlled in different output configurations.

1st output	2nd output	Linked?	Muting
A or D variable	A or D variable	Linked	Both
A or D variable	A or D variable	Separate	Independent
A or D variable	Fixed	n/a	1 only
Fixed	A or D variable	n/a	2 only
Fixed	Fixed	n/a	1 only

Balance control: with and without delay

To implement balance with delay, see page 25

601 offers a delay option on the balance control. In a conventional preamplifier the balance control simply makes an adjustment between the gain of the left and right channels. In the pre-digital era this was often important to correct for gain errors in source equipment or for imbalance in recordings.

Latterly we are tempted to use the balance control to move the stereo image from side to side. This is a very poor approach to moving a stereo image since our perception of stereo depends on time cues as much as on amplitude.

The 601 balance control can be implemented with delay - for details see page 25. When you do this you are able to correct for a non-central listening position. The balance control now delays the signal fed to the nearer speaker as well as reducing its level. You could think of the effect as rotating the sound stage towards where you are sitting off axis.

The delay implemented is correct for a listener between 1m and 2m from the speakers.

Note Because the delay is switched as you adjust the balance control it is impossible to avoid small clicks.

Do not use delay balance for the following situations. (Setting balance to 0 is enough to avoid the effects):

- When you are listening on headphones.
- When you are making a recording using that output. If a recording with delay on one channel is subsequently replayed on Mono then a 'comb-filtered' sound results.

Grounding, connections and sound quality

The 601 derives some of its superb analogue performance because the input and output are electrically isolated. For the very best listening conditions make sure that the connections you have made do not bypass this isolation by inadvertently connecting the input ground to the output ground through associated equipment.

601 is connected to the power (mains) AC ground on the analogue outputs, the digital inputs and on the electrical communications connections.

The digital inputs should be unimportant since the standard calls for the source equipment to have a 'floating', transformer-isolated output.

The analogue outputs should be unimportant since it is not conventional practise to connect the inputs of power amplifiers or active speakers to AC ground.

Some source equipment may however be connected to AC ground. Depending on the advice given by the manufacturer this may be optional. *Meridian* source equipment e.g. CD players and tuners are not grounded in this way. (An exception is the *Meridian* CD preamps 207 or 208. In these cases do not disconnect the AC ground, just connect the digital output of 207 or 208 to the input of 601. Instead of connecting a **S1-lead** between 207/8 and 601, use a **B-lead**).

Warning Under no circumstances must you disconnect the AC ground (earth) connection in a product intended to be connected this way without the written agreement of its manufacturer.

A common way in which the isolation of 601 may be bypassed comes when you connect an analogue recorder to an input and output of 601. When you are not recording disconnect the cable between 601 and the input of the recorder.

Specification

Digital inputs	32kHz, 44.1kHz or 48kHz ± 120ppm
Cable inputs: D1-4	IEC958, 75Ω
Optical input: D1, D2	EIAJ, 12MHz bandwidth receivers
Analogue inputs	
Turntable pickup MM	Nominal sensitivity is adjustable: 5mV, 4mV or 3mV for 5cm/s @ 1kHz. Input for full digital level, i.e. overload point: 47mV, 19mV or 9.5mV @ 1kHz. Cartridge load 47kΩ // 100pF.
Turntable pickup MC	Nominal sensitivity is adjustable: 600μV, 240μV or 120μV for 5cm/s @ 1kHz. Input for full digital level, i.e. overload point: 2.8mV, 1.2mV or 600μV @ 1kHz. Cartridge load 220Ω // 10nF.
High-level flat inputs: Radio, Video, Tape 1 and Tape 2	Inputs are separately adjustable. Input signal for full digital level is: 2.6V, 1V or 300mV. Input impedance 100kΩ
Digital outputs	As input for digital input sources: 32kHz, 44.1kHz or 48kHz ± 120ppm. Internal conversion 44.1kHz ± 50ppm
Cable outputs: 1, 2	IEC958, 75Ω
Optical outputs: 1, 2	EIAJ, 12MHz bandwidth transmitters
Analogue outputs	Output impedance 10Ω unbalanced, 20Ω balanced. Maximum output 8Vrms.
Fixed outputs: 1, 2	Output voltage adjustable: 450mV, 550mV, 650mV, 800mV, 900mV, 1.1V, 1.4V, 2V, 2.8V, 4V, 5V, 7V, 10V, 14V and 18V for full digital level.
Variable outputs: 1, 2	Adjustable in 99 steps of 1dB. For full-scale digital signal volume 88 gives 2Vrms @ 1kHz.
Headphone output	As 2nd analogue output. 2Vrms maximum behind 47Ω.
Digital/Analogue conversion	2 stereo 18-bit Delta-Sigma converters running with 8-times oversampling. 18-bit overall precision.
Analogue/Digital conversion	1 stereo 16-bit Delta-Sigma converter running with 64-times oversampling at 44.1kHz.
Distortion	<0.01% input to output up to digital full-scale and/or 2Vrms output.
Noise and hum	< −90dB for digital or high-level analogue inputs < −70dB MM input, < −60dB MC input.
Controls	Off, Source, DSP, Display, Mute, Volume
DSP processor	Motorola DSP56001 running at 32MHz. For details of programmes supplied see DSP section of this manual starting on page 26.
Dimensions	325 x 100 x 320mm, (12.8 x 3.9 x 12.6")
Weight	7 kg (15.5 lb.)

Help!

Standby point not lit

Check the following:

1. There is AC power connected to the socket on the rear of the 601
2. The power switch on the rear panel of the 601 is turned on

If the 601 will still not illuminate, check any fuses in your power supply and the fuse in the inlet of the 601. If these are all intact, contact your dealer.

The 601 forgets DSP settings

Every time you switch back to a **DSP** programme the 601 reloads settings for all the controls. The settings loaded are either:

- Factory preset values, or
- Values you have stored in *Setup* as described on page 31.

It does not reset to the values you last used. The reason for this is that you can, by using the special *Setup* mode, store settings that other users cannot then accidentally upset.

To go back to the factory preset settings:

1. Press **Reset** on the Remote control.

To recall your settings for the current DSP programme:

1. Press **Memory** when the 601 is in *Normal* (i.e. not *Setup*) mode.

Audible hiss at high volume settings

The input dynamic range of any recordings you have are at maximum 16 bit. The reason for this is that CD and other consumer sources use a 16 bit standard. The analogue sources you have cannot achieve this kind of range. For comparison, analogue sources are likely to be of the order of:

- FM Radio, 13 bits
- Reel-reel tape, 13 bits
- Cassette tape, 12 bits
- LP, 11 bits.

Note that there may be a difference between the dynamic range of the source channel when it is operating, and the noise you hear in standby. For example LP normally has lower noise when the stylus is not in the groove, similarly tape will be quieter when it is stopped. CD may also be quieter when it is stopped, producing so-called 'digital silence'.

The 601 has a 18 bit capability from input to output, and to provide the very best sound quality we use an advanced 'high-pass dither signal'. When the volume is turned up high you may hear this dither as a hiss when the sources are stopped. This hiss is lower than the background noise of your recordings and is of no consequence.

Poor sound quality

Poor sound quality will usually result from one of the following:

- Driving an analogue input too hard, generally this will be accompanied by a flashing of the **Level** light on the front panel. The solution is to adjust the input sensitivity for that source in *Setup*. See page 29.
- On LP in particular having the input configured for the wrong sort of cartridge. In *Config* you have the choice of moving-coil or moving-magnet types. See page 26.
- Subtle degradation may be due to ground bypass. Refer to pages 37 and 68.

Hum on LP inputs

The turntable input is the most sensitive on the 601. It is fairly common for installations to have problems with hum. The most common causes are:

- Tone-arm not connected to the technical ground of 601 as described on page 32.
- Tone-arm connected to power ground in turntable. This connection is not appropriate for 601 since the analogue input section of 601 is not connected to its output (which is connected to the power ground).
- Magnetic pick-up. If you have another piece of equipment that incorporates a mains transformer too near the turntable or the left-hand side of 601 this may be the cause of hum – particularly with MC cartridges. Re-orient the equipment to see if this fixes the problem.

Hum on other inputs

There is no reason for 601 to produce hum on analogue inputs other than LP. Check the source equipment. If 601 seems to be the cause of hum consult your dealer.

Communications not working between 601 and other Meridian products.

If you have this problem check the connections carefully. The section starting on page 35 deals with this in some detail. Then check the following:

Are you using a **S-lead** or **P-lead** instead of a **S1-lead**?

Are you using a **Q-lead** instead of a **Q1-lead**?

Have you got optical comms input and output reversed?

If you cannot see an obvious problem, connect the comms one product at a time. If you cannot solve the difficulty refer to your dealer, but first note the software versions fitted in all of your *Meridian* items.

To find out the software version for a Meridian product:

1. Press its **Display** key while it is in *Standby*. (If it does not have a display key use a 209 Remote control or refer to the manual supplied with the product).

Note your connections carefully.

Radio interference

NOTE 601 is a digital audio and computing device which has been designed to very high standards of electromagnetic compatibility.

FCC WARNING: This equipment generates and can radiate radio frequency energy and if not installed and used correctly in accordance with our instructions may cause interference to radio communications or radio and television reception. It has been type-tested and complies with the limits set out in Subpart J, Part 15 of FCC rules for a Class B computing device. These limits are intended to provide reasonable protection against such interference in home installations.

If this equipment does cause or suffer from interference to/from radio or television reception then the following measures should be tried:

1. Reorient the receiving aerial (or antenna) or route the antenna cable of the receiver as far as possible from the 601 and its cabling.
2. Ensure that the receiver uses well-screened antenna cable.
3. Relocate the receiver with respect to the 601.
4. Connect the receiver and this product to different AC outlets.
5. If the problem persists contact your dealer.

EEC This product has been designed to comply with the limits set out in EN55013 and EN55020C.

REMEMBER to switch all units off before changing any connections.

Cleaning

The main body of the 601 is anodised aluminium. To clean the case, use a cloth barely moistened with water.

Disconnect the power cord before cleaning the unit. Ensure that no water is allowed to get inside the case and do not re-connect the power until you are certain that the 601 is completely dry.

The display panel is glass, the key pad is plastic with gold plated buttons. Use a barely damp chamois leather on these areas.

Do not use abrasive cleaners on any part of the 601.

The connectors on the rear of the 601 are gold-plated and need no cleaning if gold-plated phono plugs are used. Otherwise, we recommend that you remove and replace the plug at least once a year. A proprietary contact cleaner can be used to some advantage.

Maintenance & Service

The Meridian 600 Series of hi-fi components have been carefully designed to give years of untroubled service, and there are no user-serviceable parts inside the case, nor do the units require any form of maintenance.

In the unlikely event that your 601 fails to function correctly, it should be returned, in its original packing, to your Meridian dealer. In case of difficulty, please write to:

In the UK:
Digital Gramophone & Wireless Ltd.
14 Clifton Road, Huntingdon,
Cambridgeshire, PE18 7EJ.

Outside the UK contact the importing agent for the territory. A list of Meridian agents abroad is available from the above address.

No responsibility can be accepted for the 601 whilst in transit to the factory or an agent and customers are, therefore, advised to insure the unit.

In the USA and Canada, contact:

Meridian America Inc.,
3800 Camp Creek Parkway
Building 2400, Suite 112
Atlanta, GA 30331
Tel. (404) 344-7111
Fax (404) 346-7111

Guarantee

The *Meridian* 601 is guaranteed against defects in material and workmanship for 12 months from the date of purchase.

The guarantee is void if the 601 has been subjected to misuse, accident or negligence, or has been in any way tampered with or modified without the written authorisation of DGW Ltd.

Note that connecting anything other than the correct Network-Lead to the socket marked 600 COMMS may cause damage to the 601 which will not be covered by this guarantee. Attempted servicing by unauthorised people may invalidate this guarantee.

Labour and carriage charges are not covered unless by local agreement.

When seeking service under guarantee, proof of the date of purchase will be required.

Outside the UK, local warranty liability is restricted to equipment purchased within the territory. Our agents abroad are only under contractual obligation to service under guarantee equipment sold through them. They are entitled to make a non-refundable charge for any service carried out on other equipment.

This guarantee does not limit your statutory rights within the United Kingdom.

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