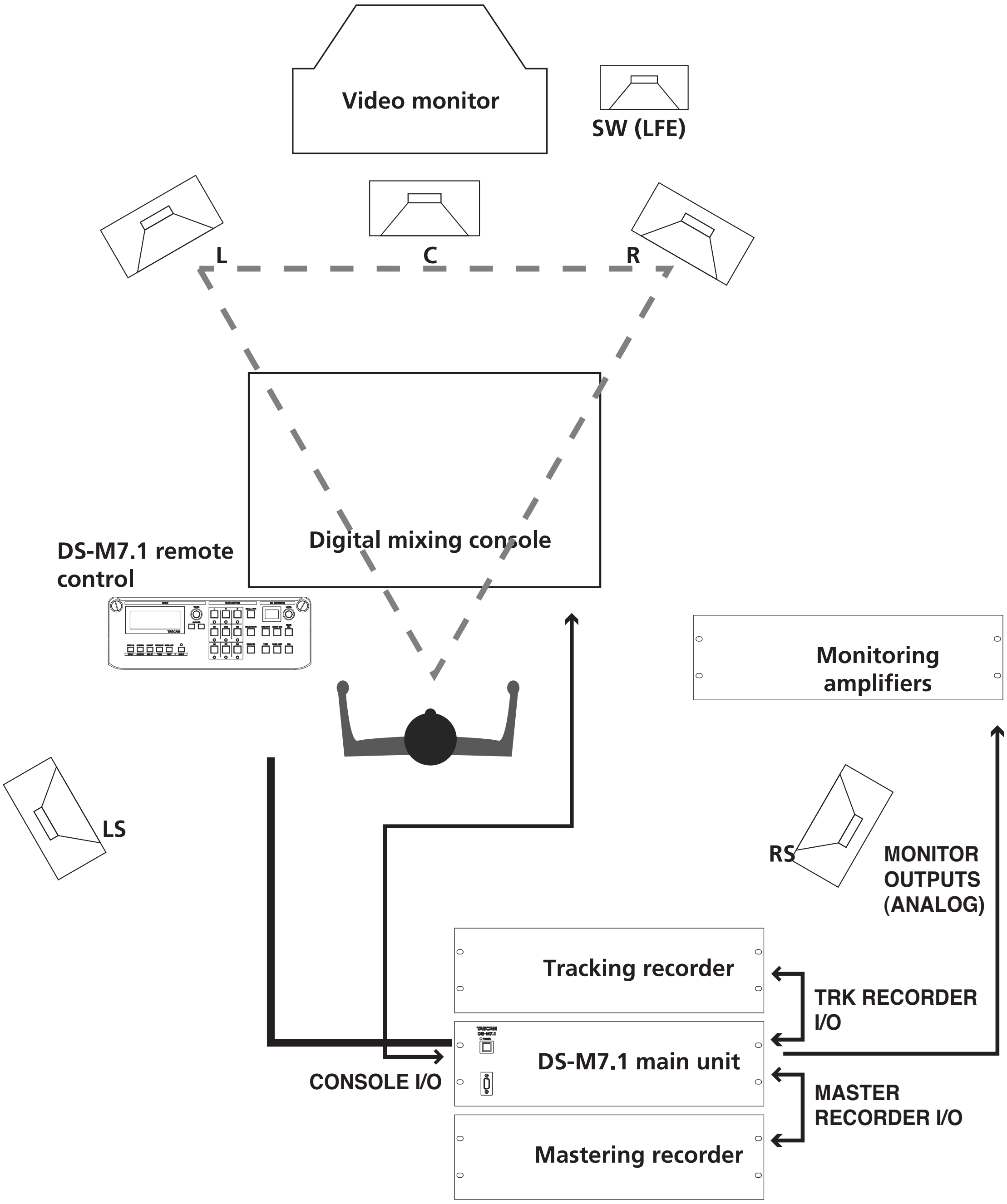
Setting up the TASCAM DS-M7.1

Before you start using the TASCAM DS-M7.1 to help you with the monitoring of your surround mix, there are a few operations to be carried out which will help you use the unit most effectively.

These are: trimming each channel so that each channel is correctly balanced relative to the others for your listening position, adjusting the delay for the channels to avoid phase delays, etc., and setting up the overall SPL (loudness) of the unit.

First, you should set up the DS-M7.1 so that your system is something like the one pictured here. For full details, see the *Reference Manual*, paying particular attention to the notes regarding word sync. Also note that this illustration shows a 5.1 system with no insert loop attached to the DS-M7.1. Your setup may differ from this, but the principles will be the same (the connections between the monitoring amp system and the monitor speakers are omitted for reasons of clarity.

Note that the connections between the master recorder and the tracking recorder are bidirectional, as are the connections between the console and the DS-M7.1.



Trimming the levels

When monitoring in surround mode, it is important that the perceived level of each channel (as perceived from the monitoring position) is equal, in order to achieve a satisfactory mix.

NOTE

Before proceeding further with these operations, it is important that you ensure that the input mode of the DS-M7.1 reflects the setup you are using and the output channels are "patched" so that the names of the channels correspond to the actual channels that you have set up in your system (if the channel meant to be driving the LFE is actually driving one of the surround channels, you will have severe problems!). Use the **INPUT** *and* **OUTPUT** *keys (shifted* **STATUS** *and* **SYSTEM***) keys for this. See the main manual if this setup has not been done already and you are unsure of the procedures.*

The DS-M7.1 provides you with a pink noise generator to allow you to set up and calibrate your monitoring system for optimal results. You will also need an SPL (sound pressure level) meter. Absolute accuracy in this case is not of prime importance—relative levels are what are being set up here.

There are two standard weightings that are commonly used, and may be selected from most SPL meters: the 'C' weighting, providing an almost uniform response from 32 Hz to 10 kHz, and the "A" weighting, which is concentrated on the 500 Hz to 10 kHz range. For full-range music productions, use the "C" weighting.

TIP

When you make the measurements described here, either stand the SPL meter on a tripod (ideal), or hold it to one side of your body, to avoid reflections, etc. caused by your body which can affect the final results.

Generating the pink noise

Make sure that your monitoring system is turned on, and adjusted to a nominal level.

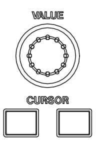
```
[TEST]
PNoise FULL BAND
Level  -10dB
Safety ENABLE
```

With the **SHIFT** indicator lit, press the **BASS MGT/TEST** key to bring up the screen at left. This allows you to select the type of pink noise generated by the unit, as well as the level of the noise.

The `Safety` parameter at the bottom of the screen, when set to `ENABLE`, turns off the ability to switch on the TEST mode and the pink noise generator. To enable the TEST mode, set the `Safety` parameter to `DISABLE`.

You will use the `FULL BAND` noise setting in most situations (the alternative is `BAND LIMIT`, which uses a filter to weight the noise).

There are a number of different selectable levels at which the noise can be output: `-20`, `-18`, `-16`, `-10`, `-5` and `0` (all relative to full-scale). If you are working to the SMPTE standard, therefore, `-20` corresponds to the analog nominal level (the EBU equivalent is `-18`—the TASCAM `-16` level is also provided for convenience).



Use the **CURSOR** keys to select the `Level` field, and the **VALUE** dial to adjust the level to the setting you want to use for your calibration.

Now press the **TEST** key on the bottom right of the control panel.

```
[TRIM]
L   0.0 C   0.0 R   0.0
        LFE 0.0
LS 0.0          RS 0.0
```

The channel **MUTE CONTROL** keys are used to select the channels to which the pink noise will be output. When an input mode is selected, the green indicators under the "active" channels are lit, to show that the channel can be used. For example, if the stereo mode is selected, only the L and R indicators are lit. In the case of 5.1, **L**, **C**, **R**, **LFE**, **LS** and **RS** are lit.

With the **SHIFT** indicator lit, press the **DOWNMIX/TRIM** key to bring up the screen at left. Note that the screen shown here represents a 5.1 mix. If you have your setup in a different configuration, what you see on screen may differ slightly.

Now, when you press one of the **MUTE CONTROL** keys, the pink noise will be output from the selected channel (at the level you set earlier). The key lights to show you which channel is currently being tested. In addition, the cursor on the screen jumps to the setting of the channel whose key has just been pressed.

TIP

When working in 5.1 mode, we suggest starting with the **C** *(dialog) channel in order to achieve best results.*

Use the SPL meter to measure the level from the channel at your listening position.

You can now press another of the **MUTE CONTROL** keys and read the level from the meter for that channel (we suggest that you do not calibrate the sub-woofer) .

Using the **VALUE** dial, adjust the level of the channel until it matches the first channel.

Repeat this process until you have a variation of not more than a few dB between channels.

NOTE

These SPL adjustments do not affect the mix going from the multitrack system to the tracking recorder—they are for monitoring purposes only.

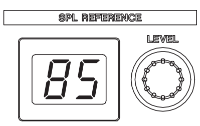
Adjusting the overall SPL

When the individual channels are balanced, you should adjust the overall level of the system so that a nominal signal generates the industry standard SPL. This is standardized at 85 dBC for movies, and 82 dBC for television work.

```
[SYSTEM]
Dimmer        -60dB
SPLRefLevel    65dB
SurroundLvl   -10dB
```

The `SPLRefLevel` menu item described here works in conjunction with the dBFS reference level of the device feeding the DS-M7.1, measuring the reference level in dBC. For example, if the source unit outputs a tone at –20 dBFS and this menu parameter is set to 85 (dBC) there is 20 dB of headroom available (as explained below). In other words, in this example the **SPL REFERENCE LEVEL** display can show up to 105 before distortion occurs. If the input signal is at full-scale, then there is no headroom available past the reference level; that is, if the **SPL REFERENCE LEVEL** is set to a value above that of the `SPLREfLevel`, distortion will occur.

Therefore, if the source program material has a dynamic range that takes it up to full-scale levels, and the standard is 85 dBC (movie), both the menu and the front panel knob should be set to 85. For other standards, the appropriate values should be set. For a project which does not use SPL standards (e.g. music projects) the `SPLRefLevel` menu item may be set to a high value (say 109) so that the **SPL REFERENCE LEVEL** control can be used as a volume control without distortion occurring.



Use the **SPL REFERENCE LEVEL** control and 2-digit display on the DS-M7.1 to adjust the SPL level in the following way:

Turn down your monitoring amplification system.

Press the **SYSTEM** key and press the cursor keys until the display shows the `SPLRefLevel` parameter. Use the **VALUE** dial to set this so that it equals the standard you will be using in your project.

Set the **SPL REFERENCE LEVEL** control so that the display shows the reference level you want to use.

Play a nominal level signal (pink noise) through your mixing console into the DS-M7.1. Take the reading of the SPL meter, and adjust your amplification system until the SPL meter shows the desired level. You may now use the **SPL REFERENCE LEVEL** control to reduce the monitoring level, but only increase it if the material permits.

Adjusting the surround level

You may want to adjust the surround level (the level of the rear speakers) so that they are anything up to 10 dB quieter than the front speakers. Use the `SYSTEM` menu above to set this value.

NOTE

The setting of the LFE crossover and relative level is handled by the dedicated bass management system, and is treated in detail in the main manual. Typically, the default settings will handle this.

Delay setting

Another important aspect of working with surround monitoring is setting up delay times, so that the signals from the different channels all arrive at the same time at your monitoring position. If this is not done properly, the resulting mix may not be quite what you expect.

The DS-M7.1 contains delay circuits to allow the monitoring to be adjusted.

For this process you will need some kind of tape measure or ruler to measure the distance between the listening position and the speakers.

Measure the distance from your listening position to the different speakers. Treat the longest such distance as the "reference" distance. The reason for this is that it is easy to "move" other speakers "further away" by introducing delay, but adding negative delay, which involves time travel, is still under development!

As a rule of thumb, one foot (30 cm) of difference equals one millisecond of delay.

Work out how many milliseconds of delay you will need for each channel.

The **L** and **R** speakers should be equidistant from the listening position. If you can arrange the **C** speaker so that it is the same distance from the listening position as the **L** and **R** speakers (that is, in an arc with the listening position at the center, then you do not need to make different adjustments for the **L**, **R** and **C** channels. If the **C** speaker is in a straight line with the **L** and **R** speakers, you will need to make the **C** speaker electronically further away, and add delay to the sound.

Likewise, the **LS** and **RS** speakers should be equidistant from the listening position (the recommended angle for these speakers is about 20° behind the listening position).

```
[DELAY]   ON
L 50.0 C 50.0 R 50.0
      LFE50.0
LS50.0        RS50.0
```

With the **SHIFT** indicator lit, press the **DELAY** key, so that you see a screen like the one at the left (this is for a 5.1 arrangement—if you have a different setup, the screen may appear different).

On the top row, move the cursor and use the **VALUE** dial to turn the delay `ON` from `OFF`.

Move the cursor to the different channels and adjust the delay time according to the values you have worked out. You can adjust the delay for each channel in 0.1 ms steps up to a maximum of 50.0 ms.

Further reading

We strongly suggest that if you are unfamiliar with surround sound techniques, or the concepts described in this guide, you obtain some of the excellent reference material available from the Internet, or reference works published on the subject.

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