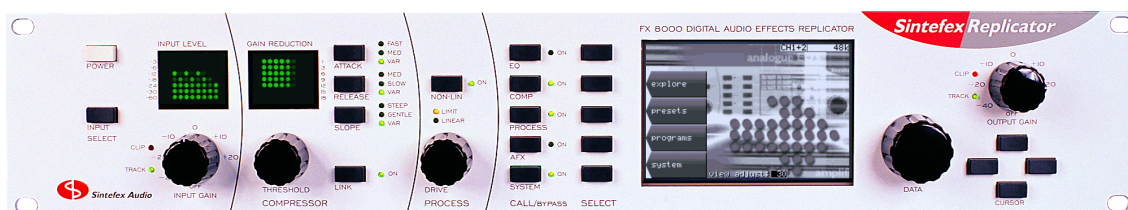




**Sintefex Audio**



# THE SINTEFEX AUDIO FX8000 REPLICATOR



## User Guide



# **SINTEFEX FX8000 REPLICATOR**

## **User Guide v3.0**

Notice: Whilst all reasonable effort has been made to ensure that the information in this manual is accurate, Sintefex Audio Lda accepts no liability for inadvertent errors, omissions or changes in operating software or hardware specification referred to in this manual.

This edition April 2003

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# 1 *Introduction*

## Welcome to the SINTEFEX FX8000 REPLICATOR.

The Replicator takes its name from the fact that it can learn the performance of analogue systems and replicate the effect digitally. This is a new process never before available to the audio engineer and opens up new possibilities. The Replicator FX8000 comes complete with many hundreds of sampled EQ's compressors, signal paths and pre-set programs.

The basic operation of the FX8000 is straightforward and you probably won't need this guide. However we recommend that after you have experimented a little you read it through to gain a better understanding of what is going on and how to make Replicator create the exact effect you need for mastering, mixing or tracking or indeed any audio processing application.

- Before you start experimenting you should read the section on installation to make sure that you operate the unit safely.

If you are looking for a unique way to change the sound of your recordings, to add warmth and colour without leaving the convenience and repeatability of digital working, then the SINTEFEX FX8000 REPLICATOR provides the answer.

## Software Versions

This User Manual is based on software version 3.0. Whilst we have made every effort to ensure that the contents are as accurate as possible, we must stress that specifications may change from those given here. Also details of operation are expected to be revised as a result of continued product improvement.

If your software version is different this guide may still be useful. All the major functions are going to pretty much the same and are likely to get easier to use rather than harder. If you have any real problems please obtain the latest version of this document from our web site wherever possible at [www.sintefex.com](http://www.sintefex.com).

Users familiar with version 2.2 software will notice the following improvements in software version 3.0:

- Program can be loaded seamlessly during program material with a choice of effect cross-fade (provided it is not necessary to change the loaded model or system configuration which requires muting).
- MIDI Parameter and Program changes are supported to allow automation of effect changes during a song, compilation or performance.
- New RMS compressor mode has been introduced to model optical compressor gain control, i.e. where the program energy controls gain reduction rather than peak level.
- Compressor gain half-wave mode has been introduced to allow creative use of dynamic second harmonic distortion rather than the purely third harmonic of full wave level detection.
- New Surround Sound compressor gain linkage options added, as well as new options for gain linkage in stereo and between groups of channels.
- External gain linkage is available for linking more than one machine to operate with common gain reduction, and for recording of gain dynamic profile.
- Improved dynamics alias rejection during gain modulation.

In addition during the life of version 2 software the following were introduced

- Stereo and Surround Reverberation: up to 48kHz sampling, not at higher sampling rates. A selection of plates, room, hall, non-linear and reverse presets are available. In addition artificial reverberation may be combined with sampled early reflections to generate previously unavailable reverberation effects. (Note however that the FX8000 is NOT a sampling reverberator but is capable of using its sampling technology for novel effects based on real early reflections.)
- Side-chain and additional main path EQ with 4 band digital EQ allowing typically up to 8 bands of EQ if required.
- Delay management has been added so that processing delays (although very short) can be matched in bypass or padded to a known delay.
- System bypass switching has been softened so that system can be bypassed without switching clicks
- Support is present for additional sample edit features when importing FX8000 v2.1 programs

- USB communications with PCs improved under certain operating systems / conditions.
- Reverse Impulse Responses: Impulse responses snapshots can be reversed in time to reverse group delay effects. This can be used for example to compensate group delay in known recording chains, e.g. magnetic tape or disk cutting paths.
- Expander Gate: An expander/gate option is now available to add to any compressor simulation, whether or not the original sampled machine had an expander or gate built in.
- Classic EQ "POWER LINEAR" (Linear Phase) mode

## Software Registration

This software is licensed for use by end-user equipment purchasers. It is free of any additional charge and is available to all end-users subject to the licence conditions.

This software is also licensed for evaluation and demonstration by authorised dealers on a limited, renewable licence.

We do ask end-users to register this software within approximately 3 months. Registration confirms your acceptance of the licence conditions and is free of charge to authorised end-users and removes all time limitations from the software. Registration may be done by mail, fax or via our web site [www.sintefex.com](http://www.sintefex.com). End users need only register a software version once - no further registration of this software release is required. See "How do I register this software" for more details.

Authorised dealers can apply for renewal licenses whenever required.

The full licence conditions are printed at the end of this user guide.

## About this Guide

You don't need a detailed explanation of every page of commands of Replicator. Most are fairly obvious and the best way to find out is to explore the controls for yourself. This guide has two main sections - an overview to get you familiar with the controls and layouts, and a "How do I?" section which takes you through the main things people ask.

This guide replaces the "Operational Reference" which was getting a bit too long-winded for general use. If you need all the details we are working on a Technical Reference manual which covers everything in detail including general maintenance and upgrade information.

# 2 Overview

## What is the SINTEFEX FX8000 REPLICATOR?

The SINTEFEX FX8000 REPLICATOR is a unique product which can capture an analogue signal path and reproduce it digitally, accurately simulating the frequency response and distortion characteristics of the original. For the first time you can

- Choose from a large library of sampled analogue EQ's, compressors and sampled signal paths to add an analogue sound without leaving the digital domain.
- Capture a favourite signal path forever for reuse at any future time
- Capture a mono process and apply it identically across multiple channels, e.g. a 5.1 surround mix
- Store favourite compressors for instant recall with a wider range of control than the original
- Get your favourite compressor sound accurately linked in stereo or surround mixes
- Work wholly digitally and get that indefinable analogue sound
- You can even switch off the non-linearity if you like the analogue sound but need it digitally clean.

Replicator goes beyond the well known process of linear convolution (at least it is well known to DSP experts even if they never let you get at it before!). Replicator's unique and patented non-linear technology takes it beyond this to reproduce the analogue characteristic of the sound changing when you drive the unit harder.

## REPLICATOR Process

The word LINEAR is used in Replicator in a special way and simply applies to a sound process in which the output follows the input exactly. If an equaliser output doubles exactly when you double the input and otherwise sounds exactly the same it is LINEAR. This is the way most digital equalisers have worked (until Replicator came along).

Most real analogue equalisers and effects do not do this and so are NON-LINEAR (NON-LIN). That's why your valve EQ sounds different when you drive it hard. This is one of the effects of analogue processing that Replicator was designed to replicate.

The heart of the SINTEFEX FX8000 REPLICATOR is a new processing system known as **DYNAMIC CONVOLUTION**. We won't go into technical details here except to say that ordinary convolution has been known for a long time but **DYNAMIC CONVOLUTION** is new. This takes into account not just the way things respond to sound at one sound level like ordinary convolution, but how they respond at a whole set of levels. This means that when you drive a REPLICATOR effect harder, it sounds different – just like an original piece of analogue equipment.

The NON-LIN button is the quick way of switching the effect of the NON-LIN operation ON and OFF.

### *Linear Phase / Power Linear vs. NON-LIN*

We also use the term LINEAR PHASE or POWER LINEAR in reference to sampled Classic EQ effects. This does not refer to the LINEAR / NON-LIN effect above, but separately indicates that the phase of the effect is linear with frequency, or to put it another way, all frequencies are delayed equally. These effects are separate and may be combined in many ways.

## How Does it Work?

Once you have a piece of analogue equipment set up to sound good Replicator takes a sample of the effect by passing a test signal through the unit and recording the result. The test signal consists of loud signals and quiet signal to detect the way the system varies when it is driven hard or not.

Replicator stores the results on an internal disk and this allows Replicator to recall the frequency response, phase response and non-linearities of the equipment at any future time, and allows you to archive the effect for as much reuse as you want.

Pre-built effects are available in a library of factory made samples of Classic EQ units and Classic Compressors. If you use these you do not need to do your own sampling.

## Main components of the REPLICATOR

### Familiarity

The SINTEFEX FX8000 REPLICATOR may be digital but it has a lot of familiar controls. We put a lot more knobs and buttons than the average digital effects unit so you can get going as quickly as possible. We even have simulated VU meters (or you can use bar meters with peak holds if you prefer).

### The Equaliser

You may be simulating a Classic EQ, or you may be running a snapshot of the last EQ you used when recording say a particular vocal overdub. This uses the PROCESS section of Replicator.

In addition, REPLICATOR always provides a digital multi-band equaliser to fine tune the sound. You can get to it with the EQ button.

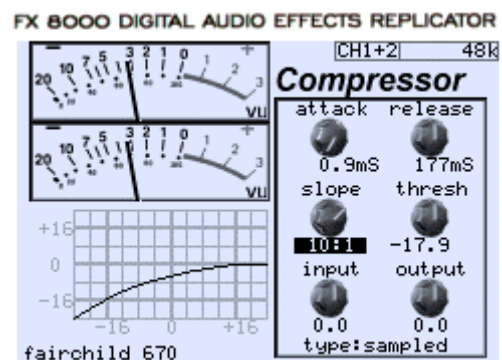
### The Compressor

Sampled REPLICATOR effects may or may not include sampled compressors. However even if a compressor is not included in the effect, the SINTEFEX FX8000 REPLICATOR unit always provides a digital compressor which is easily operated. Compressor settings are stored whenever you save settings along with the rest of the effects for recall at a later time.





Replicator's Virtual VU meters



Familiar layout of the on-screen compressor

If you are using a sampled compressor the SINTEFEX FX8000 REPLICATOR can recreate both the sound of the compressor and the way it changes with gain reduction and the 'curve' of the compressor, allowing soft knee effects to be accurately recreated.

## Replicator Process

The Replicator process itself can be configured in several ways:

- ***Snapshot simulation of a sample of a signal path***

Recreates a favourite signal path for later re-use, including the EQ settings used and simulation of any differences in the sound of the path when changing between high level signals and quiet signals. This path may include, for example, an analogue tape path, a valve (tube) device and even loudspeaker to microphone signal paths. Compression may also be included along with the compressor section mentioned above.

- ***Classic EQ simulation***

Complete EQ units which have been painstakingly modelled by merging a whole lot (usually hundreds) of samples into a dynamic, real time adjustable Classic EQ unit. We do not expect you to sample these yourself as it can take a lot of time and may need modelling a specific set of controls..

- ***Classic Compressor simulation***

Complete Dynamics Compressor units which have been modelled by sampling signal paths and dynamic characteristics into a dynamic, real time adjustable Classic Compressor unit. You can sample your own compressors but those that we have built into Classic Compressors are easy to call up and use.

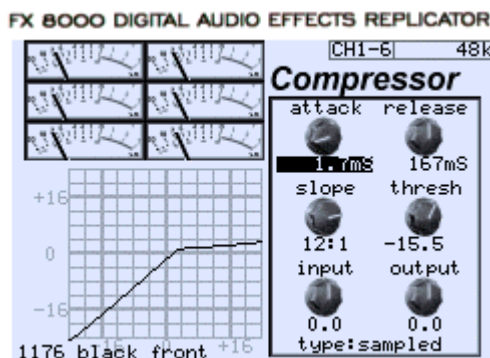
## AFX: Delays Echoes and Reverberations

AFX which stands loosely for After Effects can add a little icing on the cake of Replicated effects. A useful range of effects are available to extend the use of Replicator. For use up to 48kHz sampling rates there are a number of interesting reverberation programs for use in stereo or in multi-channel surround. Reverberations are not sampled but they can be mixed with ambient reflection samples that create some new effects.

## What of Surround Sound?

The SINTEFEX FX8000 REPLICATOR is available as a 2 channel, 4 channel, 6 channel or 8-channel unit.

A six-channel unit can support processing of 5.1 or 6.1 surround sound by applying your favourite EQ or compressor across all 6 channels (for 6.1 processing connect the 6 main channels and do not process the LFE channel). Compressor gain can be linked so that there is absolutely no image shift as the compressor operates - all channels may be locked together. In addition channels can be connected to multiple gain link buses, which may also be loosely linked together. For example the front 3 and the rear two channels can be firmly linked, and a loose linking applied between them so a big gain reduction in the front three also pulls down the rear gain by a proportionally smaller amount. See "How do I Link Compressor Gain for Special Effects?" for more details.



*Surround Sound compressor example screen shot.*

A four channel Replicator may be used for surround mixes where it is not required to process a centre or sub woofer channel. This is favoured by many music mixers.

An eight channel Replicator may be used for 7.1 surround processing, or for 5.1 surround together with a spare stereo channel (or indeed two mono channels), or for two quad surround groups, or 4 stereo groups or 8 mono groups.

Regardless of the number of channels fitted, you can always configure Replicator in combinations of 7.1, 5.1, quad, stereo and mono up to the maximum number of channels available.

## Presets: Programs, Samples and Effects

Programs, Samples and Effects in Replicator are held on an internal hard disk.

As with many music synthesisers and to allow familiar MIDI support for program selection, Each of these memory types has 8 banks each of 128 memories. (That's over 3,000 memories in total!).

### PRESETS

These are simply any program, sample or effect that has been supplied with the unit or is provided from our library on our web site.

There is a growing library of SAMPLES and PROGRAMS allocated to the top four banks of memory. In total you have memory for 1024 SAMPLES and 1024 PROGRAMS and the presets supplied either with the machine or from our growing library of effects available on the internet, provide some good starting points for your own creativity.

To save you having to move everything around as more presets become available we recommend that you keep banks 5, 6, 7 and 8 free for presets so you can download them into their designated locations.

### PROGRAMS

Whenever REPLICATOR is making a sound you like, you can save it as a **PROGRAM**. A Program remembers every setting necessary to recall that same process. Since there are 8 banks of 128 programs you should have more than enough. In addition REPLICATOR has a hidden bank of the last 128 programs you were working on when either you loaded a program or switched off the unit. You can load these using the "undo" feature of the disk save and load display and this may just save you from losing forever that sound you had just a minute ago, before you wiped it. Programs remember which **SAMPLE** or **EFFECT** you were using to make the sound.

NOTE: When you upload a program for archiving or download a program you also need the various samples that may be used to get that program sound.

### SAMPLES

When you take a sample of an audio chain REPLICATOR stores the information as one or more **SAMPLES**. These can be combined into **PROGRAMS** to get just the sound you want. (Synthesiser users may compare this with all the settings on a synth (PROGRAMS) versus the raw SAMPLES of a sound).

### EFFECTS

Effects comprise Classic EQ and Classic Compressor models which have been assembled by careful use of multiple samples and have been combined into complete **EFFECTS**. This is usually a time-consuming task

and frequently needs to have some special user interfaces developed for each **EFFECT**. For this reason **EFFECTS** appear only as factory presets. (Synthesiser users may consider this loosely like building up complex multi-sampled sounds for a music synthesiser).

### What is the advantage of 24-96 technology?

The SINTEFEX FX8000 REPLICATOR supports audio word width of 24-bits on its digital input and output and it uses 24-bit analogue converters. Working with higher word sizes means that you preserve dynamic range throughout a process and only need to "bit-reduce" at the end of an entire production process to the final distribution media.

The SINTEFEX FX8000 REPLICATOR can do over 1,000,000,000 floating point calculations per second (per 2 channels) to fully support 24-96 operation. 32-bit floating point processing is used throughout for virtually unlimited dynamic range with 24-bit precision. However when running at 96kHz Replicator has less time for processing each sample which (a) halves the number of steps in the convolution chain, and (b) halves again the duration of any convolution lengths. This affects the choice of optimum sampling rates as mentioned below.

### Should I work at 96kHz, 48kHz or 44.1kHz?

If you are working on digital material your best choice would be to use the sample rate of the material you are working on and go digitally into and out of Replicator - unless the effect you want dictates running at a lower sample rate in Replicator.

Otherwise we liken the choice to that of using 30ips or 15ips analogue tape.

- 96kHz is like 30ips, giving you a cleaner top end and allowing you to run smoothly through 20kHz up to the thirties and into the forties. This is at the expense of a poorer bottom end due to Replicators simulation "window" acting like a "tape head" of fixed width.
- 48kHz or 44.1 kHz is like 15ips; the top end is constrained to around 22 - 24kHz but the bottom end is cleaner and flatter. Of course, there is no equivalent to "tape noise" so that is not an issue.

At lower sampling rates Replicator can reproduce more ambience or stored energy in the original sample. This is a better choice for gross distortion effects or mechanical systems involving for example loudspeaker and microphone simulations.

## MIDI Control

### General

This software version features MIDI control of parameters and program loading. The system will also transmit program load and parameter control messages if desired.

Full details and a MIDI implementation chart are at the end of this guide.

### Saving MIDI setup

The settings for enabling MIDI are saved with the power-up settings of the FX8000. These settings are saved when the system is powered down or if a "save setup" is performed in the "power-up state" page.

## Installing the FX8000

### Mounting the unit and connecting AC power

The FX8000 is a standard 2U rack mount unit. It is fan cooled and accordingly does not require cooling space above or below the unit, giving the maximum packing density in a rack installation. Ventilation holes must not be obstructed and a free flow of air should be available at the sides of the unit and from the fan at the rear.

Before connecting AC power check that the voltage rating matches your AC supply. Depending on the power supply fitted and local regulations you may be able to set the voltage to the correct setting for your area, or the unit may be supplied pre-set for the area you are in.

Sintefex FX8000 Analogue Sampling Digital EQ / Compressor / Effects Unit

For optimum performance the power supply should be uninterrupted. Once connected the power switch on the rear panel (if fitted) should be left "on" and the front panel power button used subsequently for normal operation. This allows orderly shutdown of the FX2000 processes and saving of operating set-up on the FX2000's internal hard disk.

Audio and Control Connections



The connectors for unbalanced audio, balanced analysis signals, clock, optical and coaxial SPDIF are all clearly labelled. The FX8000 is supplied with multi-way break-out leads to provide access to the balance audio inputs and outputs and AES inputs and outputs. One break-out lead is required for each pair of channels.

For permanent installations the FX8000 may be wired using a standard 25-way D-type male connector. The pin-out is as follows:

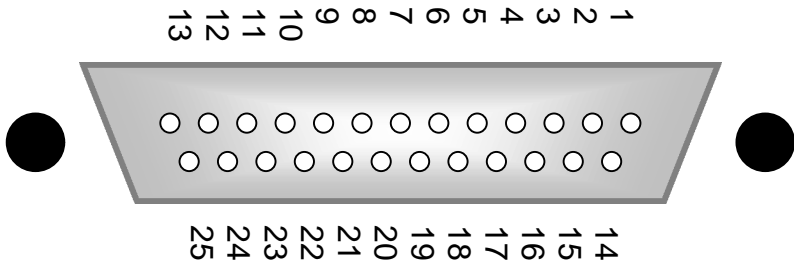
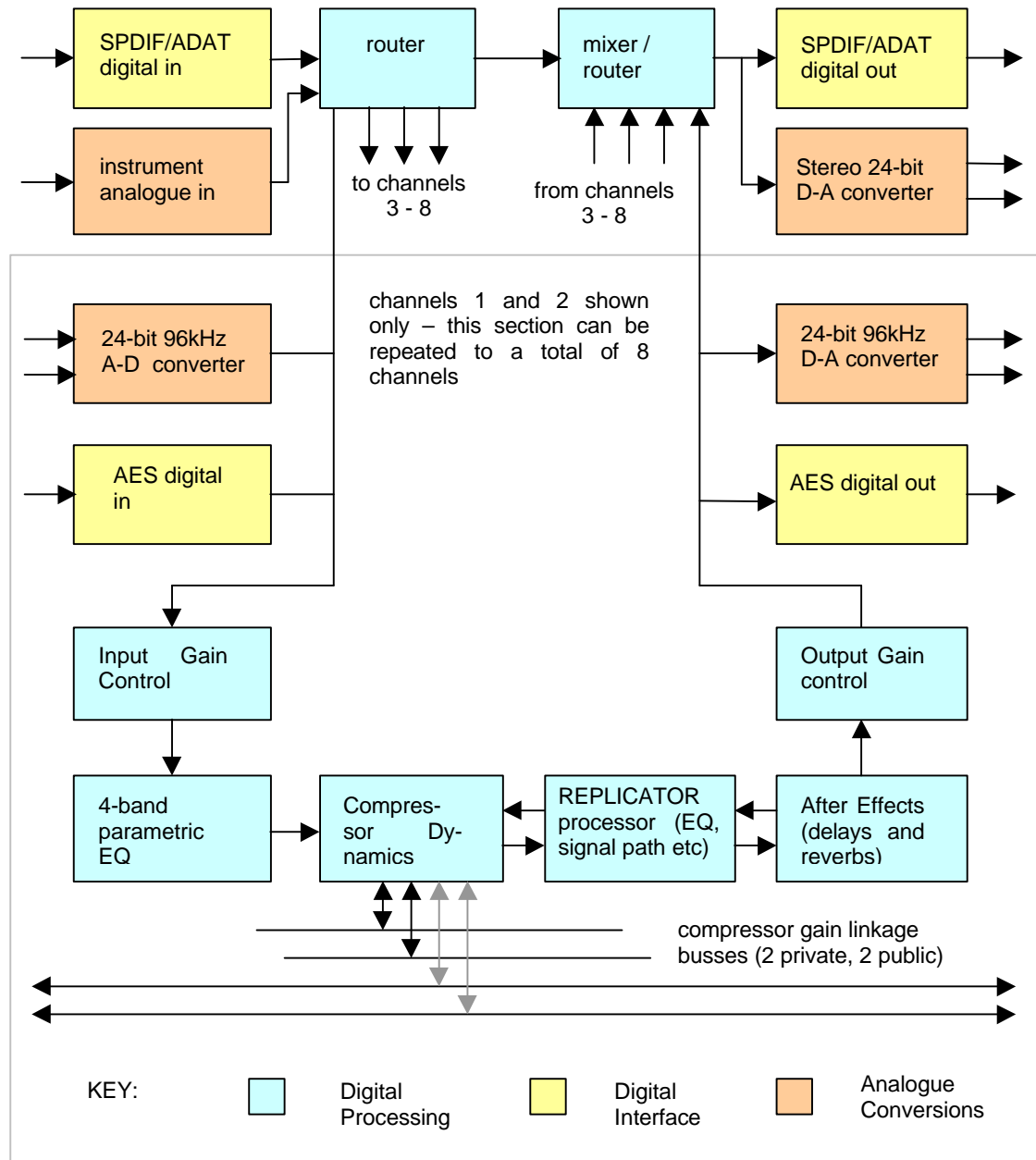


Diagram of D-type connector as seen from rear.

Each pin is connected as follows (multi-channel machines have a pair of channels on each connector as labelled, thus channels 1&2, 3&4, 5&6 etc):

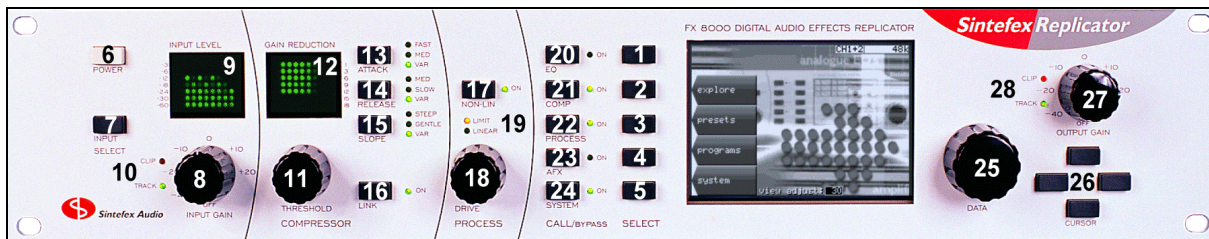
Pin #	Connection	Pin #	Connection
1	Analogue out channel 1 positive ("hot")	14	Analogue out channel 1 negative ("cold")
2	Ground	15	Ground
3	Analogue out channel 2 positive ("hot")	16	Analogue out channel 2 negative ("cold")
4	Ground	17	Ground
5	Analogue in channel 1 positive ("hot")	18	Analogue in channel 1 negative ("cold")
6	Ground	19	Ground
7	Analogue in channel 2 positive ("hot")	20	Analogue in channel 2 negative ("cold")
8	Ground	21	Ground
9	Ground	22	Ground
10	Digital AES out channels 1&2 positive ("hot")	23	Digital AES out channels 1&2 negative ("cold")
11	Ground	24	Ground
12	Digital AES in channels 1&2 positive ("hot")	25	Digital AES in channels 1&2 negative ("cold")
13	Ground		

## System Block Diagram



- It is possible to change the order of the 4 main processing elements but the diagram shows a typical arrangement.

## The Displays and Controls



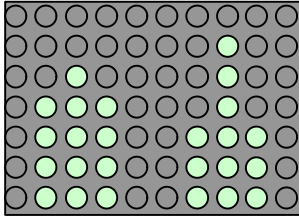
### Replicator Displays and Controls

1. Select Button 1 - selects option displayed on screen
2. Select Button 2
3. Select Button 3
4. Select Button 4
5. Select Button 5
6. Power - safe power up and down button - saves your set-up on power down and mutes analogue
7. Input Select - accesses the input / output selection screen, precision level displays and sample rate
8. Input Gain - digital input gain control
9. Input Level Display - gives confidence display of all inputs
10. Input Track and Clip LEDs - shows whether input gain is active and warns if input clips are detected
11. Compressor Threshold Control - instant access to threshold
12. Compressor Gain Reduction Display - gives confidence display of all channels gain reduction
13. Compressor Attack - quick pre-set of 3 options
14. Compressor Release - quick preset of 3 options
15. Compressor Slope - quick preset of 3 options
16. Compressor Gain Reduction Link (between channels) - links all grouped channels or pairs of mono
17. Replicator Process NON-LIN on/off - ON when simulating level dependent analogue path
18. Replicator Process DRIVE control - level control into level sensitive analogue simulation
19. Replicator Process LIMIT and LINEAR indicator LEDs - indicates if simulation is level sensitive
20. EQ in/out/call - quick press to call the display of the digital EQ, hold down to bypass/enable
21. Compressor in/out/call - quick press to call compressor display, hold down to bypass/enable
22. Replicator Process in/out/call - quick press to call analogue simulator display, hold for bypass/enable
- 221b. Where Sherlock Holmes lived
23. After Effects (AFX) in/out/call - quick press to call after effects page, hold for bypass/enable
24. System in/out (digital bypass of all processing and gain except mixing and routing)
25. Data Entry Knob - changes currently selected data on display
26. Cursor Buttons left/right/up/down - moves around values on screen
27. Output Gain - digital output gain adjust
28. Output Track and Clip LEDs - warns if output gain active or if output clips detected

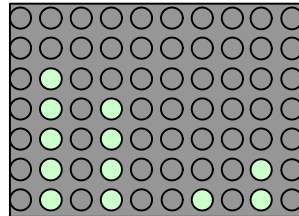


## Input Level and Gain Reduction Display

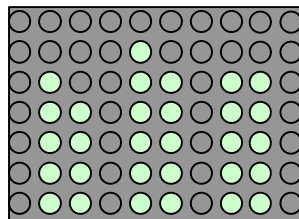
Depending on the number of channels fitted, the Input Level Display takes on different appearances:



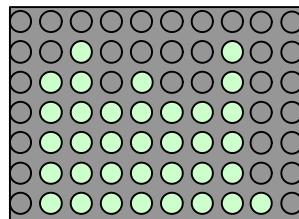
**2-channel unit**



**4-channel unit**



**6-channel unit**



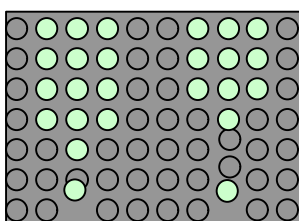
**8-channel unit**

2-channel units feature a central peak hold bar and an indicator of operating level on either side of each 3 column display. All other displays use the top LED of the bar as a peak hold indicator.

If you need higher precision displays, there are a choice of precision peak and VU meter displays available on the LCD screen. These LED bars serve as confidence indicators at all times.

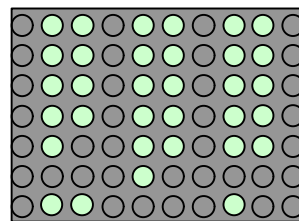
Gain reduction indication follows the same pattern, and there is a peak gain reduction hold indicator available on 2-channel units only.

In addition the bottom LED acts as an expander/gate indicator. Here is a 2 channel unit with expander/ gate active on channels 1 and 2, and a 6 channel unit with expander/gate active on channels 1,2 and 5:



**2-channel unit**

expander/gate  
indicators



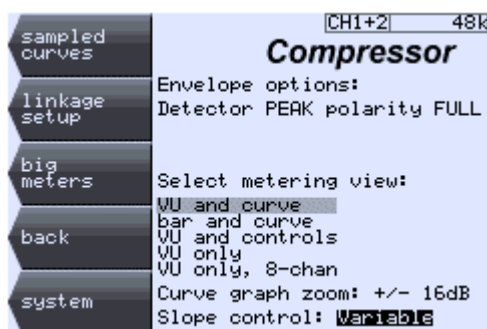
**6-channel unit**

## 3 How Do I?

### Adjust the Compressor Slope to a precise value between the steps normally provided?

Select "display setup" from the compressor page.

COMP -> display setup



Move the CURSOR DOWN to the Slope Control field at the bottom of the screen. Select the "Slope control" to "Variable".

Press "back" to return to the compressor controls page.

Compressor slope can now be adjusted in steps of 0.1 between 1:1 and 30:1.

- This page also allows selection between peak and RMS detector modes and FULL and HALF-WAVE rectification.
- The middle section of this page allows various modifications to the compressor display.
- This page also allows zooming into the knee area of curve graph (it can display plus or minus either 16dB or 4dB around the knee).

### Change the sample rate of a snapshot sample?

Whenever you load a sample that does not match the current sampling rate in force at the time of loading you get the opportunity to sample rate convert the original sample.

Sample rate conversion on loading is quick and the sample may be saved again at the new sample rate if desired (usually to a different memory location).

Sometimes samples can sound okay without conversion in which case there will be a shift in pitch of the frequency response characteristics of the sampled signal path. This may not be noticeable for sample rate changes between say 44.1kHz and 48kHz but you can try it both ways to see which effect is best.

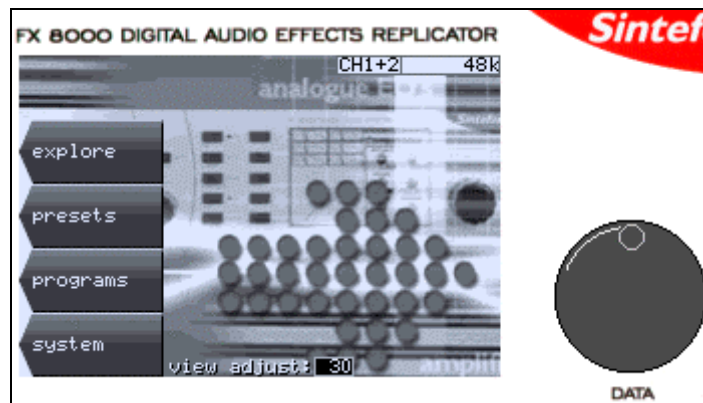
See "How do I Move a Sample or Program to a different memory?" for detailed information.

### Change the view angle of the display?

Like all LCD displays you need to adjust the view angle according to the way that Replicator is mounted, or to accommodate temperature variations.

Whenever the system is powered up the DATA knob is initially connected to the LCD view angle adjustment so an immediate adjustment can be made. (If the unit is switched on from cold you may not want to do this until the machine has warmed up as the display may be slightly dark when cold).





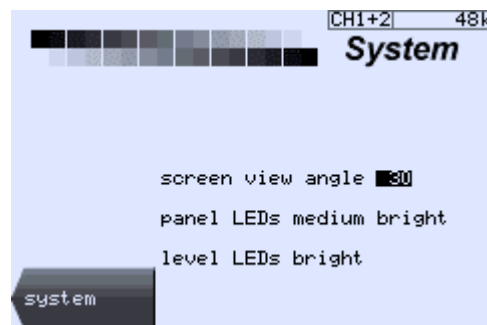
The "view adjust" field shows the adjustment number that you may become familiar with.

For adjustments later you can select the start page again

SYSTEM -> start screen

or you can go to the "screen and LED" page to adjust the view angle as follows:

SYSTEM -> setup & sample -> config -> screen and LEDs



The DATA wheel is ready to adjust the view angle or you can select 4 brightness adjustments for the panel LEDs and the Input/Gain Reduction LEDs. (LED brightness adjustment may not be available on some systems).

You can save these settings so they apply every time you start the system by selecting

SYSTEM -> setup & sample -> more setup -> power-up options -> save setup.

This saves the LCD and LED settings as well as the current choice of audio input, clock source and the other options shown on this page, which are then applied every time you switch on.

## Clear the View Angle if the screen is not viewable?

If the view angle has been adjusted beyond a usable range, it is possible to temporarily disable the software setting by restarting Replicator in its "debug" mode of operation. This is done by switching off all power and waiting 10 seconds. Then power up with SELECT BUTTON 1 held down. You get a lot of messages which we use to debug systems but within a few seconds you will be in the familiar user interface. Follow the instructions "How do I change the view angle of the display?" and save the new view angle.

This operation (of starting up in "debug" mode) is also useful to

- Find out what is going on if there is a hardware problem on start-up, or
- Prevent the system initialising the disk effect storage system if there is a problem with the file system.

## Connect up the MIDI?

The MIDI IN and MIDI OUT connections are clearly labelled on the rear panel of Replicator. These are intended for connection to a MIDI sequencer for real time control of some parameters and for program loading.

The Sequencer MIDI output should be connected to the Replicator MIDI IN, (and vice-versa).

Make sure Replicator is enabled for MIDI communications. On Replicator select

SYSTEM -> setup & sample -> more setup -> remote control.

Select the "MIDI Remote Control? yes" and press "do it".

Then go to the MIDI Setup screen and select whether you want Replicator to transmit or receive program change messages, and if you require parameter control select whether to send or receive SYSEX or NRPN parameter control messages.

For more details see the section at the end of this guide on the MIDI support.

## Connect up the USB?

Universal Serial Bus is the latest desktop bus allowing efficient connection of peripherals to PCs. It is designed to be much faster than the old serial ports common on old PCs, and more versatile than the parallel port for connecting external devices.

USB is connected using a single lead that carries data in both directions. It is connected from the PC end ("upstream") which uses a wide flat connector, towards the external device ("downstream") which uses a squarer connector.

Most PCs have a one or two USB ports and USB external devices have a single socket or a captured cable. Replicator has a socket for the squarer downstream end of the cable.

The signal can be split using a "hub" so one connection from the host can split multiple ways to many downstream external devices. Hubs can be plugged into hubs with only two limits:

1. A maximum of 5 hubs can be placed between the host and the external device. (But be careful - a host PC often has a hub built in which counts as one. meaning only 4 external hubs in series are allowed), and
2. A maximum of 126 devices can connect to a single host root port. Of course a host can have more than one root port but for now 126 devices is probably plenty.

A cable must be a maximum of 5 metres and by using five 5m cables and 4 hubs you can place the external device up to 25 metres from the host PC.

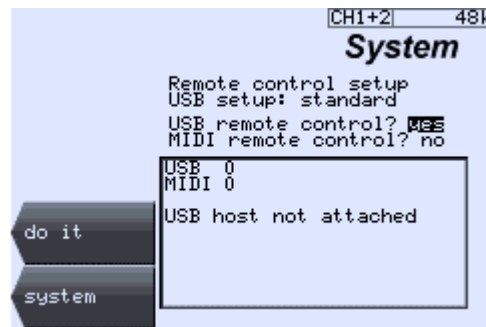
So the physical connection can be a single USB cable from the PC to Replicator, or a string of cables and hubs.

The PC must be running Windows 98 or better. Windows 95 or NT does not have USB support.

USB cables may be hot-plugged without switching off either the PC or the Replicator.

When Replicator is first attached to the PC and Replicator is set make use of USB ("USB remote control?: yes" in Replicator's remote control page). the operating system will notice the connection. Its response depends on the "USB setup:" chosen on the Replicator remote control screen.

SYSTEM -> setup & sample -> more setup -> remote control.



The preferred USB setup is "standard" and Windows will request that a disk is inserted with the appropriate hardware driver. Windows can then be told to look in the drivers folder of the Replimat software which you have downloaded from [www.sintefex.com](http://www.sintefex.com). (Other options may be available here, but their use is not covered in this manual.)

Subsequent connections will not require any user intervention unless you change the "USB setup" to a different setting.. Each time you plug in or start up Replicator the operating system notices and the connection will be made. You usually see the cursor change to an hour-glass for about half a second.

The exact steps to take are as follows.

Select the "USB setup: standard" and  
select the "USB Remote Control? yes"  
and press "do it".

Respond to any requests from Windows for drivers if this is the first connection. You will find a text document giving instructions in the Replimat directory if you need further information.

Run the REPLIMAT software on the PC and in the main menu bar select System | USB and if Replicator is successfully attached you will then be able to select "Show Input" and then "Test echo" or "Inquire" to get useful information from Replicator. You can then exit from these dialogs and Replimat is ready for file transfer or remote control.

## Edit names of programs quickly?

You can use the cursor controls to step forwards and backwards through the name of the program or sample before saving it. The data wheel changes the letter displayed. Two things that make this quicker are:

1. Turning the data wheel quickly accelerates its action so it can quickly move near the end where the capital letters and some punctuation are located.
2. If you move the cursor to a character BUT DON'T LET GO OF THE CURSOR BUTTON WHEN YOU GET THERE - turn the data wheel clockwise and the rest of the name scrolls off to the right. This way you can (a) insert some space, or (b) clear the rest of the field quickly by pushing it off the end.

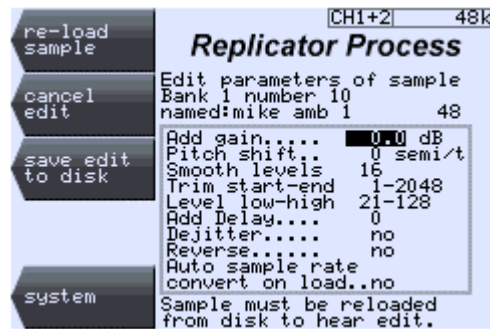
## Edit samples?

Samples are usually stored unedited on disk. When you load a sample it is processed according to the edit parameters, which are stored with it on disk and have a preset value if it is a new sample.

Assuming you have a sample loaded (use the SYSTEM -> preset -> samples page to load a sample)....

Select PROCESS -> edit sample

to get to the sample edit page.



You can vary any of these parameters but the change is not immediate. You must press "re-load sample" to hear the effect. This is non-destructive editing - the sample on disk is not altered.

At this stage the sample on disk also does not have its pre-set edit parameters changed. You do not need to do that if you save the current set-up as a PROGRAM, because the program will remember the alterations you have made.

If you want the sample always to come up with the different edit settings you have selected - for example at the very least you may have fine-tuned the "Add gain" so the sample comes up at unity gain - you can select "save edit to disk" and replace the sample with the new edit values or store it as a new sample.

The sample data on disk is not altered unless you select one of the options listed below - these settings will alter the data permanently but of course if you save it in a different location the original is unaffected.

Data is altered if

1. You have switched "Dejitter" to "yes"
2. If you switch "Auto sample rate convert on load" to "yes" and the current clock rate does not match that at which the sample was originally taken.
3. Pitch shift is not zero.

In any of these cases the data is heavily processed using a high quality sample rate conversion algorithm and at the same time the other parameters are permanently applied.

## Add Gain

This is the amount of gain adjustment to apply to the whole sample when it is loaded and should be set so that when the sample is bypassed the level on A/B is about the same.

## Pitch Shift

This re-tunes the sample for special effects, for example to change the resonance of an effect. This is similar to sample rate conversion but is specified in semi-tones so for example a resonant effect can be tuned to a particular musical effect. You will find this most effective when there is a distinct pitch present in the effect, for example a resonant peak at a given frequency.

Note that this does not alter the pitch of the signal that you are passing through Replicator. This alters the frequency response of the effect you have sampled.

## Smooth Levels

Replicator takes multiple samples of original analogue equipment at a sequence of differing levels. It is normally necessary to apply a moving average over a range of these levels to ensure that spurious noise or other effects do not generate unacceptable deviations from the original response. The default of averaging (or smoothing) over 16 level samples is suitable for many effects but this number can be increased to

- reduce non-linearity
- eliminate noise problems

## Trim start - end

Replicator normally takes impulse samples of 2048 samples duration.

Initial delay in the original sample can be trimmed off by increasing the "trim start" until the impulse response is right at the front of the displayed impulse curve. Some devices have significant delay before the peak of the

impulse and care should be taken not to trim off too much and affect the sound, but this may be a trade off between processing delay and sound quality.

If there is excessive noise or unwanted impulse data after the main response of a device, the end of the sample may be trimmed off as well. Normally this is not required.

When replicating an effect at 96kHz, the simulation length is only 1024 samples. You will find that if you trim the start to an offset greater than zero it is possible to advance the end point to a full 1024 samples beyond the start point and this is often desirable to get the maximum "window" on the impulse sampled.

## Level low - high

Replicator takes 128 samples at levels from about 40dB below peak level up to peak level (for more information see the section on sampling). This section allows you to choose a subset of these samples in the simulation.

Low level impulse response may be lost in noise, and the default value here is to discard the bottom 20 impulse responses for this reason, starting at response number 21.

- Lower this figure if the original device was very low noise and you wish to preserve low level non-linearity.
- Raise this figure if the original device was very noisy or you want only higher level signals to run into non-linearity.

If the higher level impulse responses taken by Replicator were very distorted you may want to remove them from the simulation. Setting a number less than 128 discards these top level responses.

## Add Delay

After you trim the start of the impulse, you may want to add back some delay, for example to match a real delay elsewhere in a system. This allows you to do that. However remember that the total convolution length is limited so adding a lot of delay here will reduce the active length of the dynamic convolution simulation, as you are adding "silent" impulse samples into the start of response.

## Dejitter

You may have taken a sample from a device that had variable time delay, for example an analogue tape machine where the wow and flutter results in a jittering of the sample position within Replicator's sampling window. When you load the sample the varying position of the impulse data will cause the replication process to function incorrectly.

Switch "Dejitter" to "on" and reload the sample to apply a complex de-jittering algorithm to the data. In this process, the samples are examined for to determine their peak response to an accuracy of 1/256th of a sample. The samples are then sub-sample positioned in time to an accuracy of better than 100 nanoseconds so that these points line up exactly in all the samples in the complete set of 128 samples. In addition, the samples will be phase corrected so that this peak impulse is positive in phase and located at precisely 20 samples delay in the Replicator sample window. This gives consistent delay in simulation while allowing an initial impulse response to build to the maximum as if often encountered in analogue samples.

If you have any jittering in a sample you must switch this system on to get a correct sounding sample.

## Reverse

This causes the impulse response to be loaded "backwards" or time reversed. This will have a very similar sound as the frequency response will be the same, but it will usually show:

1. Increased delay, as the peak impulse is usually moved from the beginning to the end, and
2. Time reversal of group delay characteristics.

Group Delay is the technical term for the way that most audio processors have different delays at different frequencies. This is true of all practical analogue equalisers and most other analogue signal paths like disk cutters, tape recorders etc. This spread of delay across the audio band results in smearing of transients and is often blamed for loss of stereo imaging, particularly in older recordings.

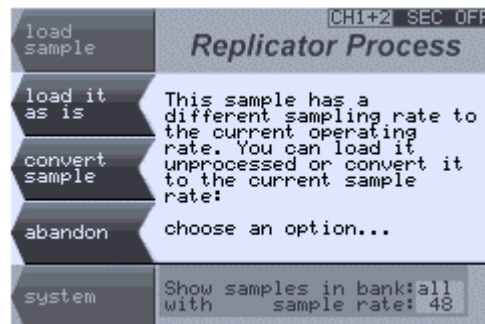
## Sintefex FX8000 Analogue Sampling Digital EQ / Compressor / Effects Unit

If it is possible to sample the original recording equipment and to reverse the response, a lot of the group delay problems can be cancelled out. In fact the frequency response is the same as the original so some extra EQ correction may be needed, but the group delay is equalised out which is otherwise almost impossible to achieve, and the overall effect can be used to restore stereo imaging precision in older recordings.

### Auto sample rate convert on load

When you load a sample that was taken at a different clock rate all frequency dependent elements of the effect will be shifted with the sample rate unless you sample rate convert. For example a sample of an equaliser with a 1kHz notch taken at 48kHz will load as a 918.75Hz notch at 44.1kHz (if you do not sample rate convert), or as a 2kHz notch at 96kHz.

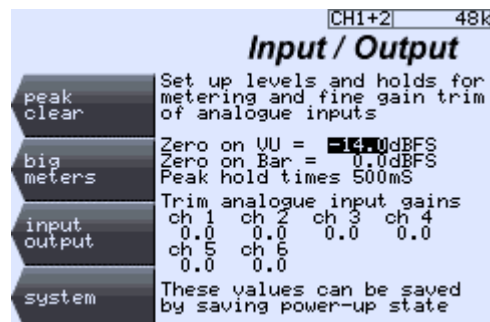
Whenever you select to load a sample taken at a different clock rate from your current one you will be asked whether you want to sample rate convert as follows:



Here you can choose the options shown. If you do not choose to sample rate convert you can use the edit page later to switch it on and reload the sample. This will then retune the effect to the correct frequencies for the clock rate you are using, so for example the 1kHz notch above will once again be accurate. Conversely if you did choose to convert you can use the edit page later to switch the conversion off again.

### Hold peaks on meters for longer than half a second?

INPUT -> levels.



Use the CURSOR DOWN key to move to the peak hold time field. You can choose between 500mS (half a second), 1 second, 4 seconds, and one month (i.e. essentially permanent hold).

Press **peak clear** to clear any peaks which are being held for a long period.

Press **input / output** to see the meters on the input page.

In this page you can also set up default levels for VU meters and bar peak meters. If you are working close to peak digital (0dBFS) a value of -4dB for zero on the VU may be appropriate. If your analogue VU meters are set for zero VU at +4dBu and the digital peak is +18dBu then a level of -14dBFS is the closest match to the analogue VU meters. However, because the digital VU meters are actually peak sensitive they will probably read higher than true analogue meters.

Finally you can digitally trim the analogue input levels separately for all the analogue inputs for precision set-up.

## Find a particular page or window?

There are quite a lot of pages for the LCD screen which control different aspects of Replicator. Quite a lot of these link to other pages which you need to get to quickly.

There are 5 main entry points to LCD pages. Four of these relate to the CALL/BYPASS buttons (excluding SYSTEM) and one to the INPUT button. A summary is:

- EQ: This displays the in-built digital EQ page.
- COMP: This displays the compressor page which controls the in-built compressor and sampled compressors.
- PROCESS: This controls the main Replicator Process of "Dynamic Convolution", and allows selecting and editing of samples or control of the Classic EQs.
- AFX: This controls the After FX which gives access to digital delays and echoes of up to 32-seconds per channel (16 seconds above 50kHz), and reverberation options
- INPUT: This allows input selection, clock selection, metering displays etc.

Note that the following button...

- SYSTEM:

...does not select an LCD page but simply bypasses or enables the whole Replicator process for the current channel or group of channels you are working on. However, the system page is an important page and can be reached from most other pages using the bottom SELECT button.

Many pages have a list of options down the left side that may be selected by pressing the matching SELECT button.

Pages that do not appear to have labels for the SELECT buttons may have "Vanishing Labels". To find out, press a SELECT button and see if a list of labels appears. If it does it gives options or allows you to move to other pages.

In this guide we often suggest a route to a particular page from a recommended starting point, often from the system page. We show these as in the following example which gets you to the program load page:

SYSTEM -> programs

Similarly to get a large VU display you follow

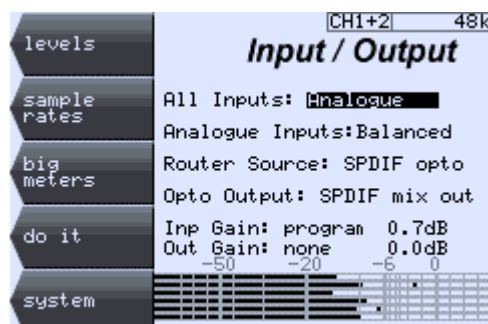
INPUT -> big meters

Some functions are a few more button pushes away, for example use the following for tidying up the effects storage system and sorting the effects into ascending order...

SYSTEM -> setup & sample -> more setup -> disk manage -> tidy disk -> sort effects.

## Fix the input and output levels so they do not change?

Select the Input / Output page using the INPUT SELECT button.



## Sintefex FX8000 Analogue Sampling Digital EQ / Compressor / Effects Unit

You will see a page like the one shown here. The input gain and the output gain may be selected to operate from the front panel control by selecting

Inp Gain: "control"  
Out Gain: "control"

If you do not want any gain adjustment on the output, and say 0.7dB gain on the input, select

Inp Gain: "program" 0.7dB  
Out Gain: "none".

When you select "program" you can move the cursor to the gain setting and choose a gain using the DATA knob.

This setting affects the current channel or group of channels you are operating. If you have a stereo Replicator you are either operating on both channels together as a stereo group or as two mono channels, in which case you can set the gain separately for each, or have one channel controlled from the front panel control and one preset. You will need to switch to the SYSTEM page to change between channels you are controlling.

When you have a channel or group selected that is set not to use from the front panel level control, the TRACK light will go out.

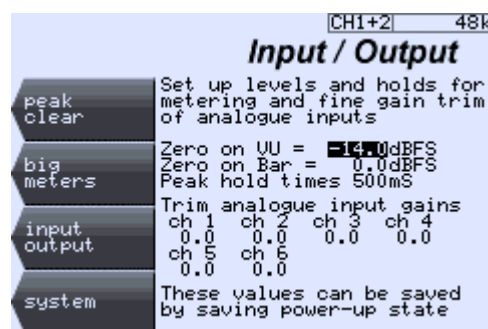
If you have a multi-channel Replicator, the process is the same but there are more combinations of channel groups available. See the section "How do I Use two mono channels on a stereo machine, or channel combinations on a multi-channel machine?" for more details of channels and grouping.

## Get Analogue Levels dead right into Replicator

Replicator is supplied with preset analogue gains into and out of its analogue converters. Unless otherwise ordered +18dBu on the balanced inputs and outputs equals peak digital audio. Other values can be supplied in 2dB steps from +10dBu to +24dBu.

All preset values can be up to a 1dB inaccurate due to variations in the converters and analogue circuitry. To allow accurate matching from your analogue environment to the digital environment the gains can be trimmed digitally after conversion and the following page shows the current settings for these trims:

INPUT -> levels.



Move the CURSOR down to the channel that you want to trim and adjust the gain after the A/D converter up to plus or minus 6dB in 0.1dB steps.

Here you can also set up default levels for VU and bar meters and set the peak hold times to a variety of values.

If your external equipment is set up so that 0VU gives +4dBu, this is 14dB below +18dBu, so you may choose the setting shown above which sets the VU simulations to show 0VU at -14dBFS. This is a good setting if you like digital headroom and your desk VU's will match Replicator's (at least on steady tones - Replicator's VU's are a bit more peak sensitive than real VU's). If however you are working near to peak digital you will probably find the VU simulations hitting the end stops all the time, so you may want to set zero on VU to (say) -4dBFS.



## Get rid of the LIMIT light or excessive distortion when replicating my effect?

When you make a sample, Replicator samples the signal path including the gain of the path. Because you often have to adjust the gain for a specific effect when you make a sample it can end up with a sample that reduces (or increases) the gain of a signal that passes through it.

If you find yourself turning up the DRIVE (which is simply a gain control into the sampled effect) to "make up" the missing gain you may not get the effect you wanted. Because samples are level dependent (they can sound different as you turn up the gain like an analogue processor often does) turning up the DRIVE is turning up the feed into the sampled effect and can result in running into distortion. In fact when the signal gets to a level through the simulation which equals the loudest impulse passed through the original equipment, Replicator has to LIMIT the signal as it has no information about how the original device would have responded above this level.

If you switch the NON-LIN off you cannot get limiting as you remove the level sensitivity, but you also lose some of the level dependent characteristics of the original process you are simulating.

The solution is to reduce the DRIVE to avoid the distortion and to add gain after the sample. If you are in a hurry just lift the output gain of the whole device, or if you want to finely adjust the gain of the sample for future use you can edit the gain of the sample in the following page:

PROCESS -> edit sample

Select the Add gain field in the edit box and increase it by a few dB. Then reload the sample and it will include the new gain setting.

See "How do I Edit samples?" for information on editing samples and resaving them if desired.

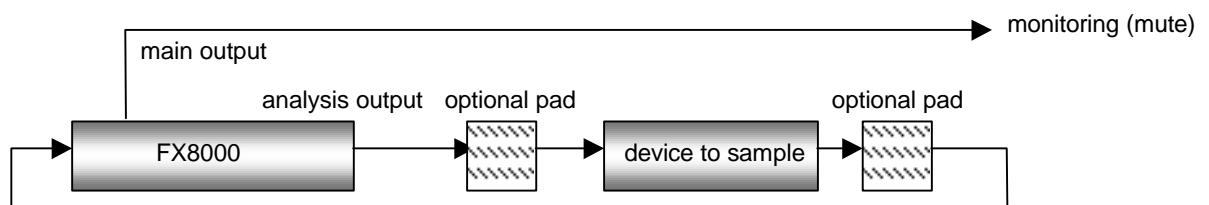
You can tell when you have the amount of additional gain right by bypassing the PROCESS using the PROCESS button (remember you have to wait for the LED to flash to toggle the effect). As you turn the sample on and off there should be no significant gain change. This will of course be program dependent when the sample has a significantly non-flat response.

## Get the level right through an effect to be sampled?

### Connect up the analysis path

Connect up the signal path that is to be sampled (see "How do I Sample a signal processing chain?").

In summary the signal path is like this...

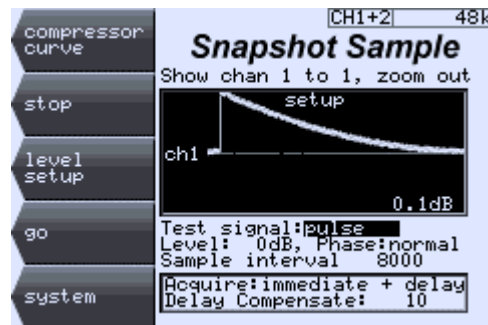


Turn any monitoring down until you have levels set up. It is then useful to monitor the FX8000 main output quietly to hear the effect of the test pulses - you can become familiar with signs of excessive distortion for instance.

### Generate the setup level tone

Select the sampling level setup page as follows:

SYSTEM -> setup & sample -> sample -> level setup.



The example screen shows the typical results of sampling a "transparent" piece of equipment. The waveform will vary according to the effect of the system being sampled.

If there is any automatic gain control in the system being sampled this must be switched off for the signal analysis, as it depends on varying level pulses being accurately assessed. Set a compressor to 1:1 or raise its threshold and keep an eye on its gain reduction. Gain reduction is sampled separately (see "How do I sample compressor curves?" for details).

When an effect is sampled Replicator sends a sequence of pulse-like signals through the device. The first of these pulses is at the peak level and Replicator expects to get a peak level response back. If the signal is any less the stored sample will have a reduced gain.

However there are a number of reasons that you need to adjust the gain through the device to be sampled. When the peak signal is being sent the processor to be sampled needs to be driven to its peak. This often means that any input level control needs to be adjusted. If there is no input level adjustment it is possible to reduce the analysis signal out of Replicator but this should be avoided wherever possible as it reduces the accuracy of the lower level samples. (It is better to use an external pad if one is available). If you need to change the analysis output level of Replicator you need to press "level setup" again to restart the pulse stream at the new level.

It is acceptable to allow the peak signal to overload slightly the device being sampled as the Replicator will then allow the effect of slight overload to be simulated.

When you have the greatest level you can push through the external device, you need to get this back into Replicator so that (a) Replicator is not clipping, and (b) you are as close as possible to peak input to Replicator.

If the device to be tested has an output gain control this can be set with Replicator's input gain set to 0dB so that the input clip red LED is just on the point of lighting.

If you want to drive the external device as hard as possible you may find that the return input to Replicator shows input clip (red LED illuminated). If one is available we suggest introducing a 6dB or 12dB pad after the device being sampled but before feeding back into Replicator. This will allow, for example, a tube output stage to drive to say +24dBu or higher without overloading Replicator - and this is a sound that can be useful to sample as often these devices have a characteristic sound when the output stage is working hard. You may also like to be sure that you have a 600 ohm termination present if there is an output transformer. A transformer without a termination resistor produces severe low frequency errors which of course Replicator will reproduce if this is what it sees. If this is the sound you are after of course, sample it the way that sounds best!

You can now make up the exact gain by increasing the input gain of Replicator until the level display on the display is as close to 0dB as possible.

The gain of the sample may well be frequency dependent so you may still have to make adjustments later when loading the sample. This can be done by editing the "add gain" option of the sample in the

PROCESS -> edit sample

page.

## Link Compressor Gain for Special Effects

Stereo Gain linking is straightforward in normal use. Simply select a stereo pair of channels and press the LINK button to link or unlink the gains. As you can see on the meters, the channel gains are linked so the same gain is applied on both channels, preventing the stereo image shifting.

Mono channels can also be linked in stereo pairs this way, letting you use two different settings or even different compressor models for the left and right of a signal, but still allowing you to link the gains. To do this both channels must have their LINK button set to ON (with the LED lit). (But see note "Matching Attack and Release Times" below).

### More About Simple Gain Linking

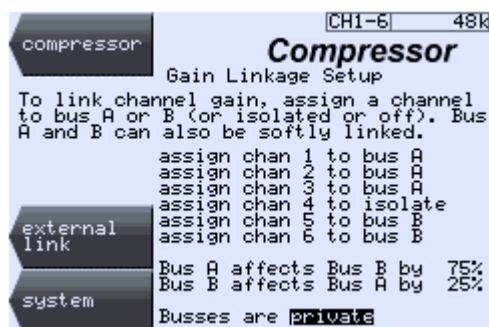
Compressor gain reduction is linked so that the channel that would generate the most gain reduction has its gain reduction applied to both channels. This means that a very loud sound appearing, say, on the left, will push the gain down on both channels resulting a drop of level on the right. This keeps the centre of the image exactly located. It is possible to reduce the effect of gain linkage without turning it off completely using the new *public* gain busses. More about this below...

New modes of linking are available from the Compressor "display & setup" page, using the "linkage setup" button.

### Surround Sound Gain Linking

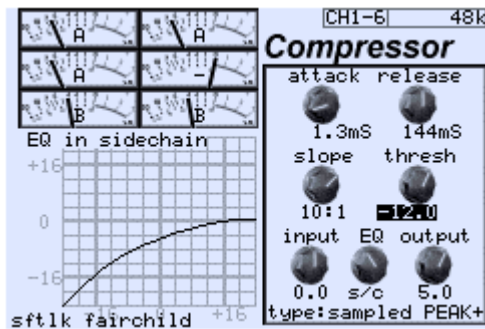
Surround Sound groups of 4, 6 or 8 channels have their gain linked in a more powerful way to allow you to separately link say the front 3 channels of a surround mix (left, centre and right) and the rear channels (left rear and right rear, and possibly rear centre).

Each surround sound group has an A and a B link bus. To start with, all the channels are assigned to the A bus so will be firmly linked together if LINK is switched on. However, to link the front 3 channels (say on channels 1, 2 and 3) together and the rear 2 (say on channels 5 and 6) together, simply connect the first three channels to the A bus and the rear pair to the B bus, like this:



You can see that channel 4 is set to "isolated" from the gain link so will not apply its gain reduction to the other channels.

When the compressor page is displayed, you can see which bus each channel is linked to: the meter is labelled A or B, and is labelled '-'. Here is an example:



Another option may be that you do not want the Low Frequency Effects channel, or even perhaps the centre dialogue channels, to have any gain reduction at all. Simply set it to "off". In this setting its gain is not affected at all but it shares the sound of passing through the same signal path as all the other channels for consistency across a surround mix. If it is off an 'X' is displayed in the meter area.

Now if you want some proportional control from say the front bus to the rear bus, you can add a percentage effect from the A bus to the B bus. If you set this to 50% then a gain reduction of say 4dB on the front bus will generate a gain reduction of 2dB on the rear pair (of course this will only be applied when the rear channels are not already achieving at least 2dB of compression for themselves). You can independently set a proportion from the rear to the front as well.

The screen above shows this example with channels 1, 2 and 3 linked on the A bus, channel 4 (LFE) isolated (but still being independently compressed), channels 5 and 6 (rear) linked on the B bus, and a 75% linkage from the front channels to the rear pair, but only a 25% linkage from the rear to the front.

## Public and Private Surround Gain Busses

The A and B busses are normally private to a surround group. However you can select the busses to be public. This allows gain linking between multiple groups. For example two 4 channel surround groups can be linked together, or a surround group can be linked to a stereo group as described below. (But see note "Matching Attack and Release Times" below).

Also the public gain bus can be linked to other Replicator machines. See "How do I Link Compressor Gain Between More Than One Replicator?" for more information.

## Public Gain Busses and Stereo or Mono

There is one public A and B gain linkage bus that is available to surround groups as well as stereo groups or mono channels. All channels linked to the public A bus will have the same gain, as will all channels linked to the public B bus, regardless of which group they are in (provided that LINK is switched on for that channel or group of course).

Stereo and mono groups that are not selected to public linkage are linked normally as stereo pairs whenever the LINK button is switched on - as described at the top of this section.

The public A and B bus can also be linked together with a percentage mix of gain reduction, so it is possible to use these on a single stereo channel pair for, say, a 50% gain linkage between channels, or they can be used to softly link a rear stereo pair to a front stereo pair or surround group on a different group of channels with a different effect. (But see note "Matching Attack and Release Times" below).

The proportional linkage on the public busses may be set from any group that is selected to the public bus, but this value will affect all the groups selected to use the public bus together.

This gives almost complete control over gain linkage for many creative effects in surround or stereo use, or even for using some audio channels to modulate others for special effects.

## Matching Attack and Release Times

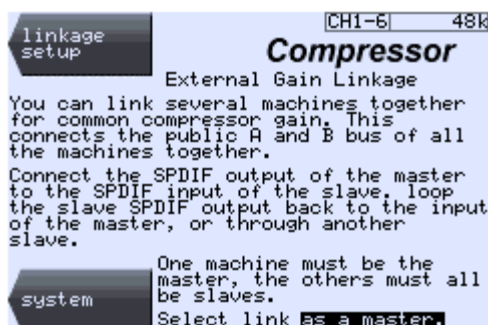
When the compressor gain linkage busses are used for linking between different groups of channels, the gain linked is subject to the final attack and release times of each different group of channels or each mono channel. Within each group of channels these are guaranteed to be all the same so channel gain is exactly matched. Between different groups or different mono channels you will need to set the same attack and release times on each channel to ensure exact stereo matching. Alternatively you can set them differently for special effects, for example where you are using one signal to modulate another.

## Link Compressor Gain Between More Than One Replicator?

The FX8000 is now very flexible in the way that gain is linked between channels. In particular there is a "public" pair of gain linkage busses. See "How do I Link Compressor Gain for Special Effects?" for more on this.

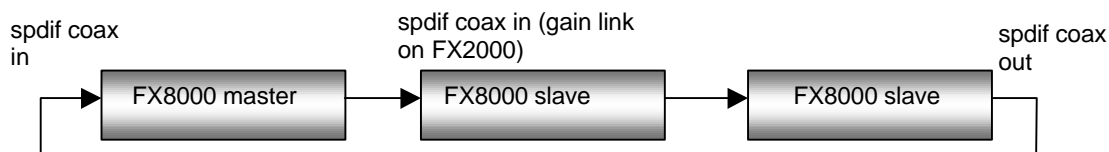
This pair of busses can be made public to more than one Replicator by use of the SPDIF connection on the Replicator FX8000 (provided of course you are not using this for audio!).

If you select "external link" on the Gain Linkage Setup page, you will see the following page:



Here you can see that you can select the FX8000 external gain linkage to be off, or for the machine to a master or slave.

If you have two or more Replicator machines you may connect them in a loop for gain linkage.



The master SPDIF output connects to the first slave SPDIF input. This machine SPDIF output then connects back to the master SPDIF input, or it may be linked to a third Replicator SPDIF input. The final slave must have its SPDIF output linked back to the master.

Select one of the machines to be a master and the rest to be slaves. Now any channel you select to use the public A and B bus will link with any other channels on all the machines.

- The COAXIAL SPDIF connector must be used - this code does not support optical connection (this leaves the optical available for ADAT connection of course).
- The compressor is still subject to the attack and release controls in each machine. For proper matching make sure these are set to the same settings.
- At least one channel of must be selected to use a public group bus on every machine for the group linkage to work. If no channel is connected to the public bus the link will not work.

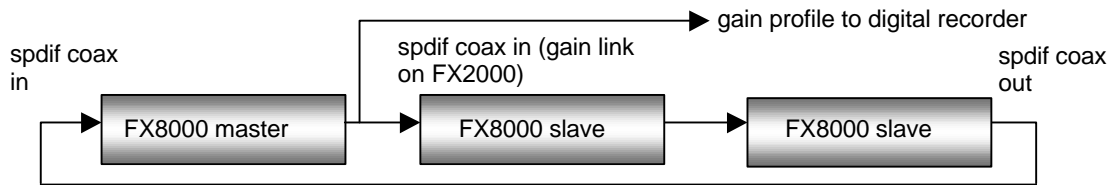
### Caution: Proper Clock Selection

It is important that all the Replicator systems that are linked together are also digitally synchronised. This will be the case in most digital environments.

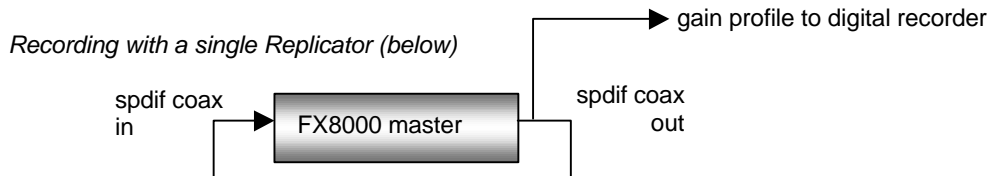
However if you are working "all analogue" you will need to make sure the machines are locked to each other. Select the SPDIF as the clock source in the clock select menu of all the slave machines.

### Recording the gain profile

A single Replicator can be selected as a master, provided that the SPDIF output is connected back into its own input. This has the benefit that if the signal is split and fed to a digital audio recorder, the gain data can be recorded. If multiple Replicators are linked the signal can be split off at any point in the loop.

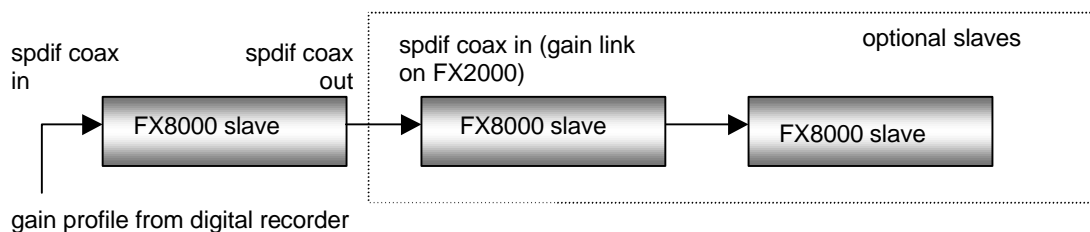


*Recording with multiple Replicators (above)*



*Recording with a single Replicator (below)*

It is then possible to "play back" the gain profile into a machine (or a set of machines). If all the machines are now set to "slave" mode the recorded gain data will cause the recorded gain reduction to be applied to the bus.



## An Application for Recording Gain Data

A typical situation where this method of recording gain reduction profiles is invaluable is where it is necessary to provide sub-mixes, say for film soundtrack work. The requirement is as follows:

- A full soundtrack is mixed in 5.1 surround, and the music producer sets the correct amount of dynamic gain reduction for the entire mix.
- It is then necessary to supply multiple parts of the mix for film dubbing each with only some of the instruments included, for example, rhythm track, orchestra, vocals (with fx), lead instruments (with fx). These are intended to be replayed together with equal level to recreate the original mix, but allows the film director to rebalance parts if necessary during the final film dub.

Normally it is not possible to do this so that if all the parts are mixed together in the dubbing suite with equal levels the same overall result is achieved as the original mix, because the dynamics compressor performs differently when instruments are removed.

Using the gain profile recording method described, a first pass is made with all the instruments and the gain profile is recorded to a spare pair of tracks of the digital recorder (in some cases only one track is required).

Then, for each sub mix, the gain profile is replayed so the exact gain profile of the full mix is applied to each sub mix. This ensures that the sub mixes will all add together to re-create the original mix as desired by the music producer and engineer.

- Note that when replaying data this completely overrides any internally generated gain.
- When replaying the gain profile, the compressor is still subject to the attack and release controls. For proper matching make sure these are set correctly.

## Gain Data Format

The gain data format is proprietary to Sintefex Audio but may be recorded on any digital recorder proved that it does not alter the signal in any way. The format uses 24-bits of data, but the method described above will work fine with a 16-bit recorder. However, the signal path between Replicators must be a full 24-bit path - do not break into it except as described above.

The data is flagged as normal audio so that it will not cause problems to the digital recorder. The A-bus is carried in the left channel and the B-bus in the right. If you are only using one of these busses you will only need one track, otherwise you will need two.

Avoid monitoring the gain reduction signal. The signal (if treated as audio) will have a lot of DC and bass content and if fed to monitors may cause damage. In particular transients in the gain profile may create loud impulsive signals.

- The gain signal cannot be recorded on an analogue recorder.
- The gain signal cannot be recorded if fed through a sample rate converter or other processor. Note that some DAW's have a sample rate converter chip fitted as standard on all inputs. These will need to be switched off.
- The gain signal cannot be recorded on a digital system that uses perceptual coding (MP3, minidisk etc).

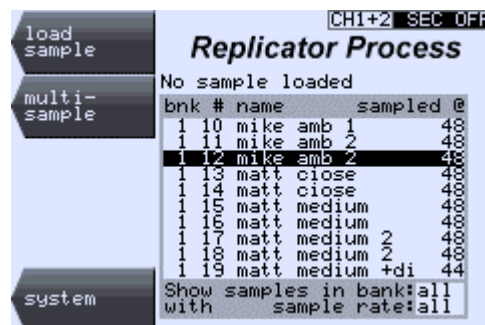
## Listen to a Sample?

Replicator has one section per channel which runs the full Replication PROCESS. To load a sample into it select

SYSTEM -> presets -> samples

If the process section is already in use you will have to confirm that you want to change to sample simulation.

You will then see the sample choice screen that can look like this:



The internal disk can contain up to 1024 samples so to limit the displayed choice the two areas at the bottom

"Show samples in bank all" and  
"with sample rate all"

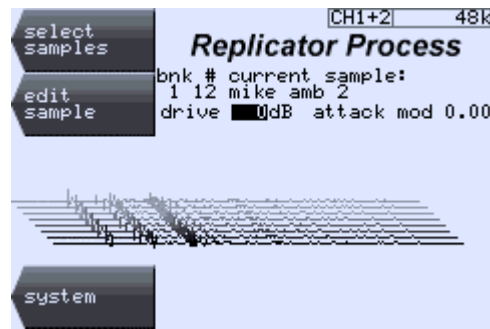
can be set to show only one of the 8 banks and only one of the 4 main sample rates. To do this use the CURSOR DOWN button to move down to the field then use the DATA knob to scroll through the selection options.

If you want to load a library sample they are usually in banks 7 or 8. Samples you make yourself are usually stored in banks 1 to 4. The screen above shows a number of samples made at two different sample rates and also show that the name you give samples can be helpful if you choose a sensible name.

After you have selected a bank and sample rate use CURSOR UP to return to the main page area and scroll through to find the sample you want to hear. When you have the sample you want to hear highlighted press "load sample".

If the sample rate for the sample you chose is correct for the current sample rate, the sample will load and you will see a screen like this:





In addition the sample processor will be switched on and (provided the system is not bypassed) you can pass a signal through the machine and hear the effect of the sample.

Two things you can vary on this screen are "drive" and "attack mod" which may be varied by moving the CURSOR to the relevant field and turning the DATA knob.

"drive" is the gain before the sample. If the sample is reducing the signal significantly, you should make up the gain after the sample using the output gain control. The DRIVE will increase the gain before the sample but will likely lead to distortion as you can only drive the sample to peak digital level. Turning up the drive will usually result in limiting taking place. You can later fine tune the gain of the sample using the "edit sample" option.

"attack mod" is an advanced option which allows some modification of the way the simulation takes place. It allows the level selection process that monitors the incoming signal to assess the level dynamically to progressively change to an envelope detection system. This can be used mainly with gross distortions to get some interesting effects.

- NOTE: If the sample loaded was from a source that may contain jitter, you must go to the edit page and switch the de-jitter option on, then reload the sample.

To edit the sample see and "How do I edit samples?".

For an explanation of "multi-sample" see "How do I Use different samples on different channels?"

## Load New Software or Presets from USB (or MIDI)?

Sintefex Audio makes new software and presets available on its web site. These are downloadable from the internet using your normal browser or using an FTP file transfer program and are ready for transfer to Replicator.

Files may be of the following forms:

1. .RFX files: These are new Replicator Effects
2. .RSM files: These are Replicator Samples
3. .RPG files: These are Replicator Programs and contain the complete setup of Replicator for a particular effect.
4. .ZIP files. These are described below
5. .LDR files\*: These are new software releases that may be transferred to Replicator using REPLIMAT

For a summary of all file types commonly used see the general information section at the end of this guide.

Files can be transferred to Replicator using a PC running Sintefex Audio REPLIMAT software, which is available free of charge to all Replicator users.

\*Loader (LDR) files and Replimat software are usually stored in a zipped file (.ZIP) for transfer over the internet. These are zipped using commercial zipping software for two reasons:

1. The files are the smallest that can store the necessary information, and
2. The files are checked by the unzipping function to ensure they have not become corrupted in transmission.

Use commercially available unzipping programs to get the contents of the zipped files. Usually you unzip the Replimat software into a folder called "replimat" and the Replicator software into a folder called "replicator". These names are up to you but we recommend you keep them separate.

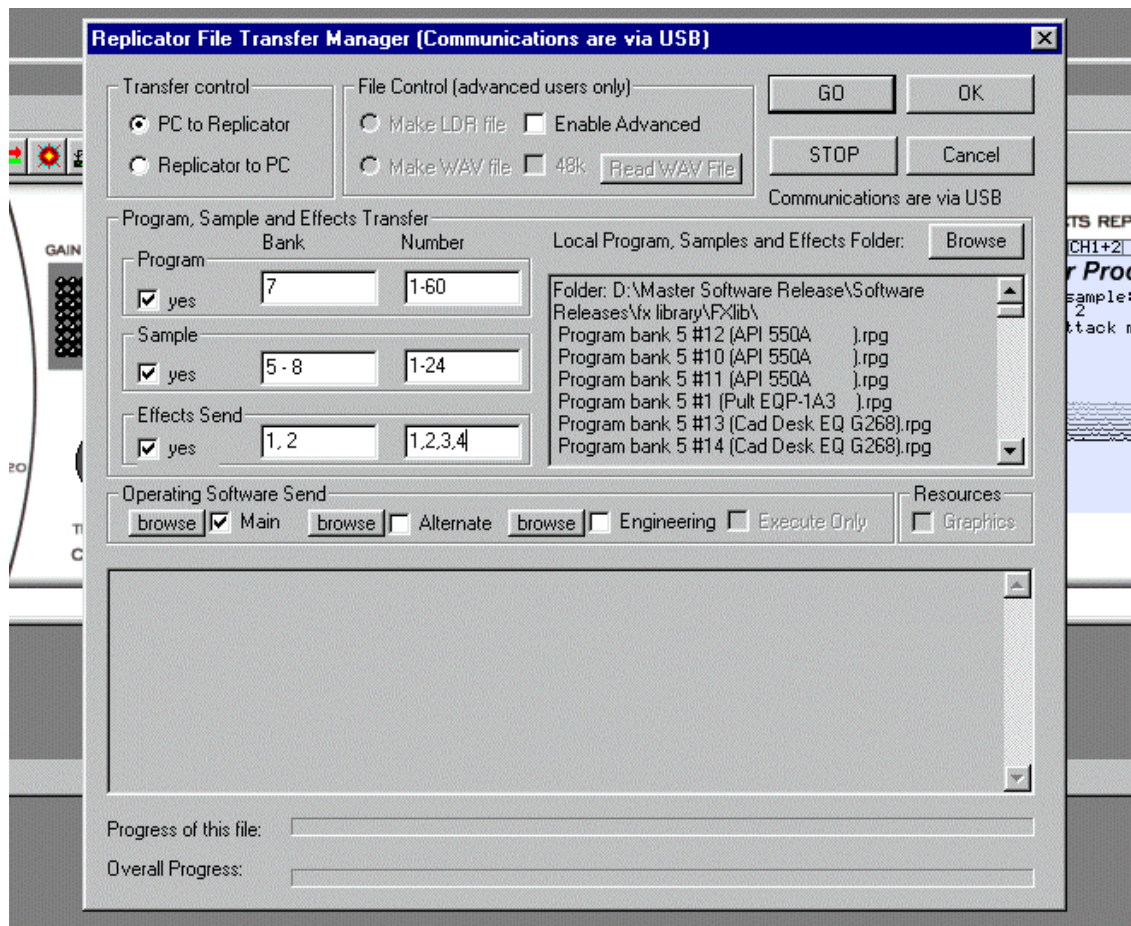


To use REPLIMAT on a PC equipped with Windows XP, Windows 98(SE) or better and a USB (or MIDI) port see "How do I Connect up the USB?", or see "How do I Connect up the MIDI?".

Once you have made a connection with Replicator, select the REPLIMAT File Transfer Manager by pressing the toolbar with this starburst legend. It looks like this...



The File Transfer Manager looks like this:



It is now possible to select the sort of transfer you want to do, for example PC to Replicator.

### To send Programs, Samples and Effects to Replicator:

1. Select "PC to Replicator",
2. Using the top right "Browse" button find a folder on your PC which contains all the programs, samples and effects you want to transfer. Select any file in that folder to select the folder.
3. Tick the require boxes for Program, Sample or Effect Send and enter the bank or range of banks and number or range of numbers (e.g. 1,2,4-8) that you want to transfer.
4. Make sure anything you do not want to transfer is not ticked - e.g. in "Operating Software Send" make sure nothing is ticked unless you are updating the Replicator software.
5. Finally press "GO". If you get a message saying that connection cannot be made, see "How do I Connect up the USB?", or "How do I Connect up the MIDI?".
6. Wait for the transfer to complete, or go and have a cup of tea.

### To send Operating Software to Replicator

- You should have at least Replimat version 2.1 for correct operation.

1. Select "PC to Replicator",
2. Make sure the boxes for sending Programs, Samples and Effects are unticked.
3. Tick "Main" to update the main Operating Software on Replicator. Click on the browse button and browse to a folder which contains the .LDR file containing the desired software to send. (This can be the same folder as used for other things).
4. It is recommended that you send only one of the three operating software at a time and that you do not send any programs samples or effects at the same time.
5. After loading the software, restart Replicator so that the new software can take effect.

### Uploading Samples and Programs for archiving

1. Select "Replicator to PC",
2. Using the top right "Browse" button find a folder on your PC into which to uploads all the programs, samples and effects you want to transfer..
3. Tick the require boxes for Program, Sample or Effect Send and enter the bank or range of banks and number or range of numbers (e.g. 1,2,4-8) that you want to transfer.
4. Make sure anything you do not want to transfer is not ticked
5. Finally press "GO". If you get a message saying that connection cannot be made, see "How do I Connect up the USB?", or "How do I Connect up the MIDI?".
6. Wait for the transfer to complete, or go and have a really strong cup of tea.

You can't upload Effects or Operating Software in case you were wondering.

### Advance Options

#### *To send Graphics to Replicator*

Graphics Resources just means things that aren't built into the software but used by it. At the moment this is just the graphic images that Replicator uses. From version 2.0 software these are built into the .LDR file supplied as the new software release. However if there are any problems with the graphic images on Replicator after the software is updated you can send a fresh version of the Graphics from Replimat as follows (you should have the latest Replimat for correct results - check our web site.).

1. Select "PC to Replicator",
2. Tick "Enable Advanced" to allow for this special option,
3. Make sure the boxes for sending Programs, Samples and Effects are unticked.
4. Tick the "Resources" Graphics box to send the latest graphics to Replicator.

### *Making Waves*

This is an obsolescent function that allows a WAV file to be created containing coded Replicator software or downloads. It is not recommended that this method of Replicator update is used now.

### *Other items*

In case you are wondering what this button is for...



This button is the RECALL button for any "report box" that Replimat sometimes shows.

Report Boxes are boxes in which Replimat puts occasional useful information. It does not stop you carrying on (you can ignore it without having to press "Okay"). After a report box has been on the screen for few seconds it goes away of its own accord.

The recall button brings back the last report box if you missed it or want to read it again. If there have been a number of report boxes recently every time you press RECALL you go back to an older one.

At the moment these are used only by the WAV file reader but in future this will be used for other useful reports.

### Renaming PC files

Programs and samples are recognised by REPLIMAT by the first section of their name, e.g:

program bank 7 #3 (tight delays ).rpg  
sample bank 8 #15(ls to mic 1m ).rsm

If you requested to load a program bank 2 number 7, REPLIMAT will transfer the first occurrence of a file matching the name

program bank 2 #7\*.rpg

where the \* matches any name.

You can therefore change the bank and number of a program or a sample by changing the file name. The information in brackets is optional and is put in by REPLIMAT from the name stored with the sample or program to help you identify it.

Effects are different and although we use the convention, for example

fxb1p014.rfx

to indicate the effect for bank 1 number 14, the name can be anything provided it has the .rfx suffix. The bank and program number are embedded in the file and cannot be changed.

REPLIMAT will download the first file with extension .rfx it encounters which has the correct bank and number embedded in it, regardless of name.

### Several versions of uploaded files

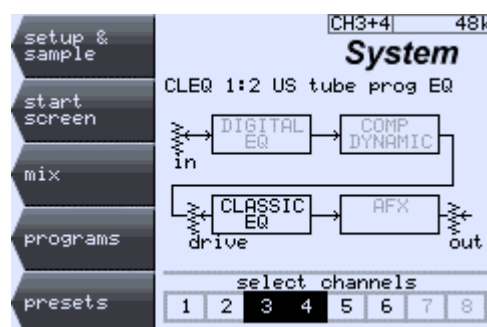
When REPLIMAT uploads programs or samples from a Replicator to a PC it looks to see if the specified bank and number has already been uploaded. If it has, the upload continues but REPLIMAT adds a (1), (2), (3) etc to the end of the file name to indicate that it is a newer version of the uploaded program or sample. You may want to delete or rename older versions so that that do not get downloaded again in error.

## Make the controls work - they do not seem to affect what I am listening to?

You are probably listening to a channel that is not the one selected on the front panel. The controls only affect the currently selected channel or group of channels (if they are grouped together). If you don't know about grouping channels, see "How do I Use Two Mono Channels on a stereo machine, or channel combinations on a multi-channel machine?".

The input and output level controls usually affect all channels, unless they are set to a fixed input and output gain. (See "How do I fix the input and output levels so they do not change on an effect I have fine adjusted?").

Select the System page which looks like this:



The data knob selects amongst the groups or channels available which will be highlighted in black. The picture shows channels 3 & 4 selected as a stereo pair.

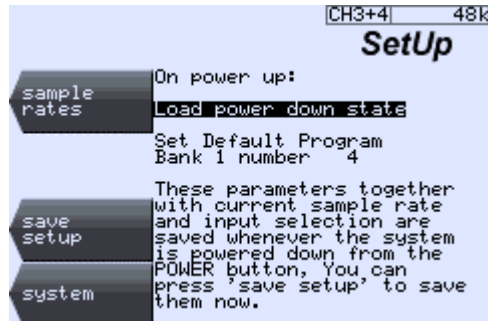
The system graphic shows that of the 4 main processing block, only the Classic EQ is used for this program. The name of the last loaded file is shown at the top.

Select the channel or group you need to adjust. You can then return to the page you are working on and the controls should be working for this group.

## Make Replicator start up the way I want?

SYSTEM -> setup & sample -> more setup -> power-up options

provides a page which allows you to make certain decisions about power-up.



The selection shown "Load power down state" is the most common. Replicator always saves its last operating state when you press the POWER off button at the end of a session. The next time you power up after setting this option the close down operating state is restored.

If the power is interrupted other than by the power down button, the state is not saved but Replicator does occasionally store an "undo" state whenever certain operations are performed. The last such state will be loaded at the following power-up.

Other options are "Don't load anything" which prevents Replicator from loading any previous state at power up. You may want this if

- you like a clean start at every power-up, preferring to load a saved program by hand
- you habitually get many power outages so shut down saves are rarely performed, or
- you power off the whole setup centrally without using the power down button.

Finally you have the option "Load the default program". This always loads the program shown below it at power up and is useful if you like to always start with a known state, or if for example Replicator is in a permanent live installation with the same starting setup every day. You can of course choose the bank and memory number for the program to load.

## What is Saved and Restored on Power-up

The following is saved at power down and restored at power up before any other operation is performed

1. The choice about what to load on power up described above
2. The clock rate of the system
3. The clock source (whether internal, reference, digital input, or automatic)
4. The audio source for each channel (analogue, digital, balanced, unbalanced etc)
5. LCD view angle setting
6. Meter level settings for 0dBVU
7. Peak meter hold time
8. Fine gain adjustments of digital inputs.
9. USB and MIDI remote control options.

## Move a Sample or Program to a different memory?

### To move a sample:

1. Go to the sample load page (SYSTEM -> presets -> samples) and choose and load the sample you want to move
2. Go to the "save" page (PROCESS -> edit sample -> save edit to disk)
3. You may want to rename the sample in the name field at the bottom of the screen
4. Select the bank and sample number where you want to save the sample.
5. If you have different samples in different channels you can select whether you want to save a single sample, for example "Save samples from:ch 1" or saving a group of samples from a multiple sam-

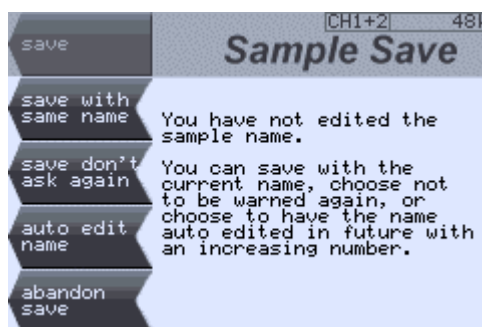


ple set-up "Save samples from:all". If you choose more than one a sequence of memories will be highlighted for you to verify that the destination locations are free.

6. Press "save" to perform the save to disk.

If one of the memories you will be writing to is not empty you will be asked to confirm the save. You can also select never to be asked this question again which is useful if you are deliberately overwriting a large selection of samples. (This is reset at next power on).

If you have not edited the sample name you will see the following screen...



The options give you the following choices:

1. Do the save with the unedited name
2. Do the save with the unedited name and do not present this screen again (until reset by the next power down)
3. Append a 3 digit number automatically in the last 3 characters of the sample name. This is incremented on every future save and you will not be prompted with the query again (until reset by the next power down)
4. Abandon the save - this returns to the save screen and allows you to manually edit the name or abandon doing it altogether.

The automatic numbering is useful when you are saving a large number of items .

Thing to note...

- The original sample remains at the original location so you actually have a new copy of it
- Any programs that use this sample will continue to use the old copy unless you edit those programs or create new ones using the new sample.
- You only get the chance to save multiple samples in one go if you have a multiple sample program set up. See "How do I Use different samples on different channels?".

### ***"Sample has not been reloaded..."***

If, when you select "save" you get a message like the one above it is because you have made changes to the edit values displayed but you have not reloaded the sample, so you are not listening to the sample the way the currently displayed edit values indicate.

If you want to save what you are listening to, confirm the save. If you want to save the edited version, reload the sample first, make sure you are happy with the sound of it, then select "save".

### **To move a program:**

1. Load the program you want to copy to a new location. (SYSTEM -> programs). If you want to save the information for all channels to a new location load the program by going into the "more options" and select "load whole system". This will load all the channels on the system not just the one(s) you are working on.
2. Select "save" to get a display of saved programs and empty memory locations.
3. Scroll the an empty memory location in the list, or one you want to overwrite.
4. Edit the name if you want to.
5. Press "save program" and a copy of the program is made at the new location. (You will have to confirm that you want to overwrite an existing program if the memory is in use).

## Undo / Redo

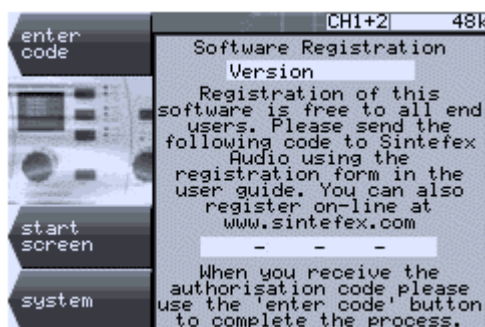
When you load a program, the previous state of Replicator is saved as a program in the undo memory. Pressing "undo" steps back through the undo memories, and "redo" steps forward again. There are 128 levels of undo.

Undo saves are also done when the power is switched off using the front panel power button (but not when power is removed or Replicator is turned off with power switch at the rear on the PSU when fitted).

## Register This Software?

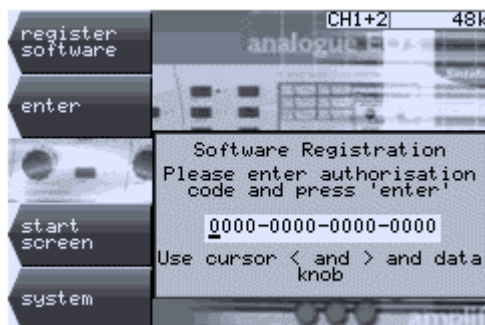
After installation this software will operate normally for approximately 3 months (720 hours) of use. At any time during this period authorised end users may register this software which will remove any time limitation.

From the start screen, select the "Register Software" option. You will see this screen which displays a code unique to your system.



Please copy this unique code to the registration form at the end of this manual. Add the software version number (also shown on this screen) and the other information requested. Then either mail or fax it to Sintefex Audio - the contact information is shown on the form. Alternatively you can register this software on-line at [www.sintefex.com](http://www.sintefex.com).

Sintefex Audio will then issue an authorisation code. This should be entered by pressing the "enter code" button in the page shown above. You will then see this page:



Enter the authorisation code in the four 4-character fields shown by rotating the data knob to select a character, using the cursor left and right keys to move between characters. Once the code is correct press the "enter" button to complete the process.

- The code is not case sensitive so you can use lower case instead of upper case if you wish.
- The letter O can be used instead of the number 0 to avoid excessive eye strain!

Provided you have entered the code correctly you will see a confirming message. If the message reports that there is an error in the code, simply check it against the code you have received from Sintefex Audio and correct any errors.

After the code has been entered correctly you will not see the "register" option again and you will have unlimited further use of the software.

Authorised dealers and those evaluating the system can renew the trial period by supplying the same unique code to Sintefex Audio with details of the requirement. Sintefex can issue a renewal code which is entered like

a full authorisation code and will cause the evaluation period to restart. Note the same code cannot be used again, a new unique identifier code will be created for future authorisation.

## Reload Replicator Software from scratch?

This is a very unlikely occurrence as you usually load Replicator software a little at a time, adding to your existing effects, programs or samples and updating software releases from time to time.

However you may need to do this if

- You replace the system disk, or
- If you have an unexpected system failure that damages the disk contents (very rare).

Note: A system failure is pretty rare and even if you have one you are unlikely to lose the programs and samples stored on the disk as these are stored in a fail-safe filing system. However you can lose effects so these may have to be reloaded after reloading the software. If you have to change the disk you will of course lose everything on the old disk.

If the system disk is replaced by an authorised representative it will be returned to you loaded with the latest Replicator software and all the latest effects library. You may need to reload your own effects afterwards using Replimat - see "How do I Load New Software or Presets from USB or MIDI?"

Sintefex Audio can also supply a replacement disk preloaded with the standard release software to authorised technicians.

The steps for reloading software from scratch depend on which start ROM you have fitted. To determine which, power up the system with the 4th SELECT button held down (2nd up from bottom). This causes the start-up process to stop and wait for user interaction rather than try and load software from disk.

Typical displays are for example

START ROM MAIN MENU v0.6.2-0 *(on the top line)*

or

System Start (version 1.0.0-6) *(on the line after the title line)*

The shows that you have start ROM version 0.6 or 1.0 respectively. See the relevant instructions below.

If you have any problems with any of the following steps, please consult customer support.

### Steps for reloading software (ROM version 0.6)

1. Make sure you have a PC with a MIDI interface available, and that you have the latest copy of REPLIMAT, the Sintefex Audio Replicator remote access program running on the PC. You also need a copy of the Engineering Software (usually called en\*.ldr - where the \* could be any sequence of digits identifying the version - for example en1-0-01.ldr or engapp.ldr) and the Replicator software (usually called r\*.ldr, for example rm2-1-00.ldr or repapp.ldr). (You can skip the engineering software but not the Replicator software).
2. Connect up the MIDI connection from the PC. See "How do I Connect up the MIDI?" for details.
3. Ensure that Replicator is waiting in its start-up software with the message "START ROM MAIN MENU" displayed at the top of the screen. If this is not the case, power down Replicator, then re-power it while holding down the 4th SELECT button from the top (2nd one up from the bottom).
4. In REPLIMAT select MIDI as the mode of communication with Replicator (the start ROM does not understand USB or WAV files).
5. Tick the PC to Replicator box and tick only "Engineering software" and browse to the folder that it is in. If you do not have the engineering software or you have any problems with USB in the next few steps go to step 8.
6. Press "GO" and wait for the engineering software to be sent.
7. After Replicator says "saved eng s/w", you can enter the engineering software by powering down Replicator for 10 seconds, then powering up with the 3rd SELECT button held down (the middle SELECT button). Replicator should show an engineering test window. Select "remote" and Replicator is now ready for full communication using USB or MIDI. If you have USB ready to go, select Replimat to USB (see "How do I Connect up the USB?") and proceed to send Replicator software using USB - it is faster than MIDI.

## Sintefex FX8000 Analogue Sampling Digital EQ / Compressor / Effects Unit

8. Tick the PC to Replicator box and tick only "Main software" and browse to the folder where the LDR file is kept.
9. Press "GO" and wait for the Replicator software to be sent.
10. When Replimat says that the transfer is finished, Replicator should be ready to operate again. Power off for 10 seconds then power up normally. You should see the familiar screen.
11. Look through the samples and programs to see if you need to load any from the PC. Try this again for the Effects. Load up any necessary presets and you are done.

**Steps for reloading software (ROM version 1.0 or greater)**

1. Make sure you have a PC with a USB interface available, and that you have the latest copy of REPLIMAT, the Sintefex Audio Replicator remote access program running on the PC. You must have Replicator software (usually called r\*.ldr, for example rm2-1-00.ldr or repapp.ldr). You also may want to have a copy of the Engineering Software (usually called en\*.ldr - where the \* could be any sequence of digits identifying the version - for example en1-0-01.ldr or engapp.ldr). (You can skip the engineering software but not the Replicator software).
2. Connect up the USB connection from the PC. See "How do I Connect up the USB?" for details.
3. Ensure that Replicator is waiting in its start-up software with the message "key 1 = remote, 2 = test" displayed on the screen. If this is not the case, power down Replicator, then re-power it while holding down the 4th SELECT button from the top (2nd one up from the bottom).
4. Press the top select button to enter the remote mode. If you are properly connected to the PC you will see the message "usb ad=n cf=n n" on the display where the first 2 numbers depend on the USB address the PC operating system assigns and the configuration number (usually 2). The last number is an indicator of USB activity. If you have never loaded any drivers on the PC the operating system will ask for the driver that came with Replimat now. If the first two numbers are 0 the PC connection has not been made. Try unplugging and re-plugging the USB cable.
5. In REPLIMAT select USB as the mode of communication with Replicator.
6. Tick the PC to Replicator box and tick only "Engineering software" and browse to the folder that it is in. If you do not have the engineering software go to step 8.
7. Press "GO" and wait for the engineering software to be sent.
8. Untick the "Engineering Software" box and tick "Main software" and browse to the folder where the LDR file is kept.
9. Press "GO" and wait for the Replicator software to be sent.
10. When Replimat says that the transfer is finished, Replicator should be ready to operate again. Power off for 10 seconds then power up normally. You should see the familiar screen.
11. Look through the samples and programs to see if you need to load any from the PC. Try this again for the Effects. Load up any necessary presets and you are done.

**Sample an Analogue Tape Recorder?**

We recommend that you become familiar with sampling an audio channel (for example an EQ unit) before attempting this more advanced operation. See "How do I Sample a signal processing chain?"

**Special Considerations**

An analogue tape recorder is like any other signal processing device in that it can replay while it is recording, providing a signal path that can be sampled directly, although it differs from a signal path like an equaliser or compressors in two ways. First the delay between the recording and replay head must be accounted for, and second the wow and flutter of the system means that steps have to be taken to compensate for this.

It is also possible to make a recording of the Replicator analysis tone at one time, and then play it back later. This can be used to sample the effect of leaving a tape "on the shelf" for a period. This is like sampling a vinyl or acetate disk, or like off-line sampling. See the section "How do I sample a Vinyl or Acetate Disk" for details of this procedure.

We should mention that the analogue tape machine should be properly lined up with respect to bias and equalisation for the tape sample to be used, unless of course you want to sample the sound of a badly lined up tape recorder, which may be a worthwhile effect to sample for some applications.

**Delay**

A typical mastering machine may have 35mm (1.5 inches) between the recording head and the replay head. This means that at 15 inches per second (15 ips, about 38 cm per second) there is a delay of about one tenth of second through the machine (half this at 30ips).

The best setting for Replicator is to keep the standard sampling setting "Acquire immediate + delay" and choose for example a 4000 sample delay compensation (8000 at 96k sampling, 2000 at 30ips 48k sampling).



You need to press "level setup" each time you change the delay setting to see the effect as this restarts the sampling timers.

If the pulse test signal is now being fed through the tape machine back into Replicator, it should be possible to see the pulse signal in the usual way on Replicator's Snapshot Sampling screen. Make sure that the zoom setting is "zoom out" to see the full length of the sample. You can now fine tune the delay to bring the impulse near to the front of the window.

You can also zoom in to fine adjust the start of the sample and this shows the problem of wow and flutter referred to below. The important thing is not to adjust the delay so that any pulse signals occur before the start of the sampling window. It is quite okay if when zoomed in the pulses are barely visible at the right. When zoomed out the pulses need to occur within about the first 10% of the displayed window.

## **Wow and Flutter (Jitter)**

Because the tape speed does not remain digitally exact, the exact delay between applied sample pulses and returned pulses varies all the time. If you are fine tuning the delay setting when zoomed in you can see this happening on the Snapshot Sample screen.

The Replicator process needs to run a whole set of sample pulses through the machine it is sampling. This is to characterise the machine at high audio levels and low audio levels in order to reproduce the way the machine varies with signal level. This is of course especially important for analogue tape.

Replicator also needs to precisely align all the pulse samples taken to generate a proper simulation. Luckily Replicator has an automatic de-jitter system built in which can be applied when the sample is loaded back from storage after the sample is taken.

This is also discussed in more detail in "How do I Edit Samples?", but in summary:

Once you have the set up the levels (see below) and have taken the sample you can load the sample into the PROCESS page. Before the sample is ready for auditioning you will need to select "edit samples" and turn the "de-jitter" option to "yes". Then reload the sample. It will take a few seconds to analyse all the jitter information and to position the samples so that the peak impulse is properly aligned for every sample. This process analyses the location of the peak to within 1/500th of a sample period, then resamples the impulses digitally to this accuracy.

## **Level**

It is important to drive the test signal quite hard onto the tape. This is because when simulating the tape machine later many sounds get their characteristic analogue sound from driving the tape hard and this will need to be simulated. Replicator can only simulate up to the level you drive onto tape at the analysis stage as it has no knowledge of what a louder signal would sound like.

In fact there are benefits in taking samples at several different levels for different simulations. This would allow you to process for example drum tracks through very heavily driven tape, but a full mix through a less heavily driven tape effect. You can do that anyway using a heavily driven sample but using less drive later but if you have different samples they may need less "fine tuning" later.

## **Sample Interval**

Analogue tape is a complex electro-magnetic effect and the response to an impulse stimulus can take a little while to decay. For this reason we recommend that you set the sample interval quite high when sampling analogue tape. We suggest for example 48000 sample intervals at 88.2 and 96kHz sampling and 24000 sample intervals at 44.1 and 48kHz sampling.

## **Tape Speeds, Tape Brands etc**

Each tape machine has its characteristic sound, which also depends on tape speed and the brand of recording tape. We suggest that samples are taken for all the combinations you would normally choose to use on a session.

## **Deleting heavily distorted samples**

It is possible with some tape machines for the higher level samples to be heavily distorted by inadequate electronics in the machine. If possible the samples should be done again with a slightly lower level driven to

## Sintefex FX8000 Analogue Sampling Digital EQ / Compressor / Effects Unit

the machine or a different machine line-up to avoid this. However if this is not possible all is not lost; the offending sample can be deleted at the sample load stage by using the "Level low-high" option of the edit page. Typically the lowest level sample to be used is around sample 21 (counting from the quietest to the loudest) and the loudest available sample, number 128 is the highest. However you can set the upper number to a lower value to eliminate grossly distorted samples from the simulation.

### Setting the "Smooth" parameter

It may be better to increase the setting of the "Smooth" parameter in some cases. This is basically averages across a number of different level samples. In the event of a tape "drop out" one or more impulse sequences could "out of line" with the rest, and this averaging is effectively removes the undesirable effect of the occasional poor impulse sample. Typical values may up to half the difference between the low and the high level samples selected.

### Permanently De-Jittering Samples

Once a sample has been taken and the "edit sample" options (including de-jittering) have been adjusted for the best simulation, we recommend that the sample is saved again with these settings so that it is permanently de-jittered and ready for use. This usually includes adjusting overall gain to get a unity gain effect.

## Sample a Vinyl or Acetate Disk?

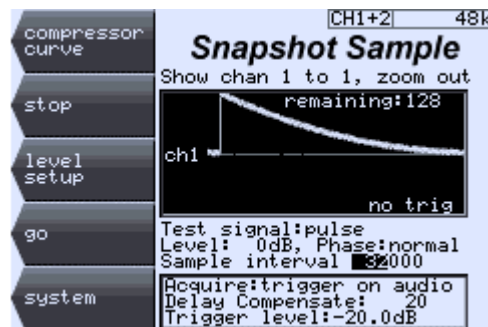
This applies also to any sample that is made "off-line", that is where the Replicator test signal is passed through any machine where the recorded response is available at a later date. See "How do I sample effects "off line"?"

We recommend that you become familiar with sampling an audio channel like for example an EQ unit before attempting this more advanced operation. See "How do I Sample a signal processing chain?"

### Special Considerations

It is first necessary to "cut" the sample analysis tone generated by Replicator onto the disk to be sampled. The acetate or a final vinyl pressing can then be played back into Replicator for sampling.

Replicator must be set to capture the effect by trigger on audio, and the sample interval must be set to match the sample interval on the original cut. The clock rate must also be the same.



The sampling screen can be set up similar to the screen shown above and then the system must be "armed" by pressing "go". Replicator will also start generating its analysis signal but this can be ignored.

The disk containing the analysis signal recorded on it can now be played. You should see the trigger occur and the impulse will appear on the screen.

If the disk is not running at exactly the correct speed, the impulse will "walk" across the screen. This must be corrected by means of a varispeed control, as once Replicator has been triggered it will expect the impulses to occur at the predefined regular intervals.

- If the disk player does not have varispeed it is also possible to varispeed Replicator by setting a user clock rate in its Sample Rates page.

By careful adjustment it should be possible to get the train of impulses to remain fairly stationary on the screen. The screen is displaying 2000 audio samples, and a variation of 10 - 20% is permissible (200 - 400 samples). On no account should any sample fall before the window or beyond say 25% of the way across.

During this adjustment you can frequently restart the disk, press "go" again to restart the counter (if you let it get to zero it will go to the sample save screen and you will have to press "back" to continue as you are not ready to save yet).

Once adjustments are complete, the disk should be stopped, Replicator should be re-armed by pressing "go", and then the pulse sequence should be started from the disk from the start of the sequence. It is important that Replicator triggers on the very first pulse of the sequence.

Watch the screen for any excessive deviation of the impulses, and finally save the sample.

It is then possible to load the sample to audition the effect. It will always be necessary to make sure that "De-jitter" is on. See "How do I edit samples?" for more on this.

## Hints

When choosing the interval between pulses to cut to the disk it may be best to use a set of different values. The further the impulses are apart, the less they may interact but the more difficult to adjust the varispeed for re-sampling.

If working at 48kHz or 44.1kHz, we suggest a sequence at 8000 samples intervals, and again at 16,000, 24,000, and 32,000 sample intervals.

If working at 96kHz or 88.2kHz, we suggest all the above plus 48,000 and 64,000 sample intervals.

It will be necessary to keep notes as to which sequence is which and what sample rate was used as this may not be obvious at a later date.

## Using a 96kHz sample sequence at 48kHz

In fact the analysis pulse sequence from Replicator at all sample rates is very similar. A 48kHz analysis sequence at 8,000 samples interval is distinguishable from a 96kHz analysis sequence with 16,000 samples interval only in the bandwidth of energy in each pulse which extends only to 24kHz at 48kHz sampling but to 48kHz at 96kHz sampling.

It is therefore possible to use a 96kHz pulse train to make 48kHz samples with the necessary adjustment to the sample interval, provided that the energy in the pulses between 24kHz and 48kHz which will not be sampled does not disrupt the system too much. This may be an issue with the disk medium which is severely limited in high frequency energy.

If a 48kHz pulse train is later sampled at 96kHz there will be no response above 24kHz in the samples as the pulse train did not contain energy in this range.

Some experimentation may be needed according to the medium to be sampled, and although there is some freedom in "mixing and matching" pulse trains and sample rates some care is needed.

## Sample compressor curves?

### Snapshot Sampling vs Compressor Sampling

#### What is the difference between Snapshot Sampling and Compressor Sampling?

Snapshot sampling takes a static snapshot of an processor and allows you to recreate its fixed effect on an audio signal. This applies to equalisers, amplifiers, microphone and speaker combinations for example.

Compressors are different. They have a definite sound of their own regardless of the gain reduction, and they also have a distinctive gain reduction characteristic. This means that they all differ in the amount of gain reduction they apply at any specific signal level. These together are what make different compressors sound so different.

## Sintefex FX8000 Analogue Sampling Digital EQ / Compressor / Effects Unit

Snapshot sampling can capture the basic sound of the compressor signal path and this is a necessary step in compressor sampling. However in addition you need to capture the gain reduction curve – and this is what we mean by sampling the compressor curve.

In fact, Replicator can capture a whole set of curves for different gain reduction slopes (also known as ratios) and allows you to use them when you are simulating the compressor. In addition Replicator can work out curves for intermediate slopes and also down to 1:1 so after sampling a compressor curve you can simulate any setting from the highest slope curve you have sampled down to 1:1.

You can also choose threshold settings after sampling so you end up with a completely programmable compressor using the curves of the compressor you sampled.

In addition you can apply the signal processing characteristic of the compressor signal path to make the simulation very close to the original sound.

- Any gate in the compressor should be switched off. Replicator is technically capable of sampling gate parameters but the current version of software will not do that and the gating effect will confuse the curve measuring system.

## Connecting Things

Connection is the same as for sampling a signal processing chain (see “How do I Sample a signal processing chain?”). The ANALYSIS OUTPUT of Replicator is connected to the input of the compressor, and the output of the compressor is connected to channel one input of Replicator. You only need to sample one channel of a stereo compressor. Replicator will replicate it across as many channels as you need later.

You can use the balanced analogue input or the unbalanced consumer level input depending on the nature of the compressor's connections. You can even use an external A to D converter as go digital into Replicator if you prefer.

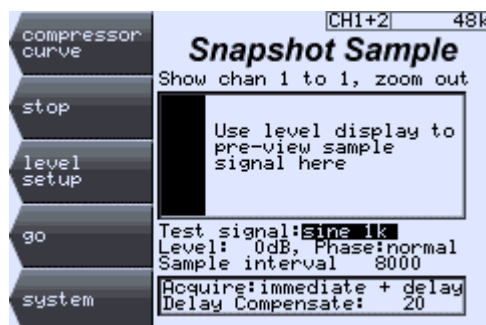
Although we are sampling the compression of the unit, you can have equalisation in the path if you want – so a complete vocal patch for instance with EQ and compression can be sampled. If you do a full audio-path sample as described later you will get the whole effect!

## Sampling Things

Select the main sampling screen

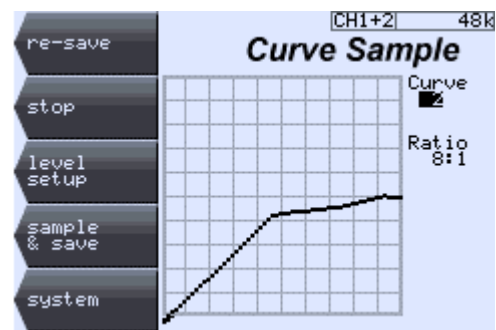
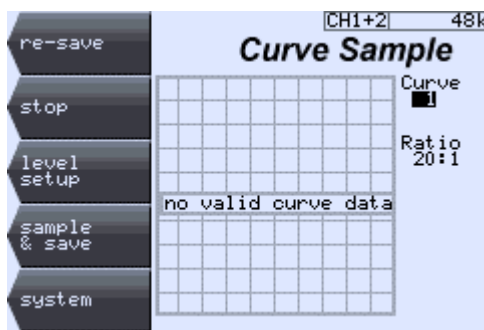
SYSTEM -> set-up & sample -> sample.

You see this screen:



Before you move on we suggest you make sure that the “Test signal:” is set to “sine 1k” as this is the most useful tone to send through a system while setting up levels, etc. The level of 0dB is right to send a peak level tone through the device while setting up levels, but keep the monitoring down. Press “level setup” to start the tone generation.

Then press “compressor curve”. You see the following screen (left hand picture):



You can use the data knob to move through all 24 compressor curve memories – they are going to be all empty at this stage unless Replicator has already been used for compressor sampling. If it has you will see something like shown on the right above.

Either way you can sample over the memory number shown after “Compressor Curve”.

It is a good idea to press “level setup”. This will generate a reference tone (as selected in the previous page) and you should be able to hear the tone going through the compressor and ending up back in Replicator. You should see the input level indicating the tone and the level is displayed at the bottom of the screen.

The intention now is to save at least one compressor gain ratio curve. One is enough if you just want to recreate the effect you have set up on the compressor but you can save a whole lot and then be able to recreate the whole range afterwards.

Choose “Compressor Curve 1” for your first sample.

We suggest that you set the input threshold on the compressor to –20dB and the compression ratio low, say 1.5:1 or 2:1. Next tell Replicator that this ratio setting will apply to the curve by selecting the Ratio field to match the compressor. If there is an output makeup gain on the compressor we suggest you turn this up so that you get about –10dB into Replicator. If not, don't worry as Replicator will figure it out for itself.

Set the compressor attack medium to fast and the release medium to slow. If there is no control try it as it is; Replicator is able to measure the curve with a wide range of settings. Switch off the test tone by pressing “stop” and wait for the gain reduction to drop to zero before doing the test.

You can now press “go” and watch while Replicator sends a sequence of tones through the compressor to map the gain curve. These will start out around –60dB and run up to 0dB so you may want to be near your monitor volume control.

At the end of the test, Replicator will save the curve in the memory and also display a graphic of the curve as shown in the picture above. If you forgot to enter the ratio – so it still says “unkn'n” (meaning unknown) you can enter the value now and press “save” which simply updates this information. You can also do this if your experience allows you to assign a more accurate ratio that labelled on the compressor, or of the compressor did not give figures.

Now don't alter threshold settings on the compressor but select another ratio and choose “Compressor Curve 2” on Replicator. Enter the same threshold setting you used before and the new ratio. Press “go” and wait again. You will know by now that it is only a minute or so.

You can repeat this for as many settings as you like. Be sure to get the steepest curve you can as this is the steepest curve you will be able to replicate. However Replicator can always go to shallow curves down to 1:1 even if you only take one curve. Some compressors only have one curve and we suggest you guess its ratio – enter say 10:1. Some have the steepest curve at 10:1 which is fine as it is pretty steep. Some go to limit (same as infinity:1) and this is worth sampling. You can get this effect on a Urei 1176 by pressing all its ratio buttons in at once! Save this under “limit”.

You have a choice now. To get the full effect you need to do the next section “Snapshot Sampling the Compressor”. However you can apply the curves you have just grabbed to Replicator's equaliser straight away by jumping on to “Simulating the Compressor Gain Curve”.

## Snapshot Sampling the Compressor

This is pretty much the same as ordinary snapshot sampling.

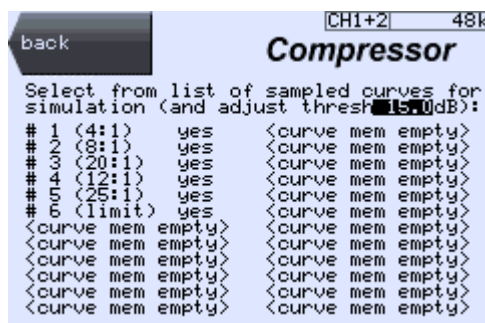
The only difference when taking a snapshot is that we want the compressor to have a fixed non varying gain during the process. Replicator passes a whole sequence of different pulses through the compressor and if these are subject to gain changes you will get excessive distortion in the simulation. You can still use them of course by switching the NON-LIN off when simulating as this gets rid of all distortion. However you will also lose the desirable distortions that are the subtle part of the way the compressor changes the sound.

Most compressors can be persuaded to give a more or less fixed gain by selecting a 1:1 setting, setting the longest attack possible, the longest release possible, and the highest threshold possible. You can make an example sample and watch the gain reduction metering on the compressor during the sample. If it moves excessively it might be better to repeat the sample at a lower level. If you still get some variation you will probably still be okay. The best thing is to press on and try it. You will probably find that by backing off the PROCESS DRIVE or selecting NON-LIN to off you will get the sound you want. (You may even improve on the original compressor sound).

Finally save the sample in an empty or unwanted sample memory for the final simulation.

## Simulating the Compressor Gain Curve

Press the COMP button to get the compressor display, and then select the "sampled curves" option. An example screen is here showing the ratio curves stored in each of the 24 memories.



Here you can see the result of taking 6 samples from a compressor at the ratios shown. You can select the curves to be used by changing between "yes" and "no" for each curve. You would do this if you had several different compressor curves stored in the same program. It also does not matter which order the curves are in. (If you mix curves from different compressors you will get some interesting hybrids!).

The field "adjust thresh 0.0dB" can be adjusted to raise or lower a sampled threshold to the centre of the simulated curve. as seen in the main compressor display page.

Go back to the compressor page (press "back") and select the "type:sampled".

You can immediately use the gain characteristics of the sampled compressor – try varying the slope and threshold. Along with the attack and release you get a lot of the characteristics of the sampled compressor.

If the "knee" of the curve is very low in the displayed window, which will also mean that the threshold you are getting is audibly much lower than the displayed threshold, you can go back to the curve select page and adjust the threshold to a higher value. Although the effect is the same as just using a different threshold setting this will mean that you can switch between simulations of different compressors and the threshold control will be consistent between them.

## Adding the Audio Processing Characteristic

To complete the compressor simulation you should press the PROCESS button to get the Replicator Process page.

If you already has a process installed you can start a new process most easily by pressing SYSTEM -> presets -> saved samples. Now you need to find the bank and number where you stored the compressor snapshot and load this into all the channels you that you want to use for compression.

This takes one or two seconds.



Finally make sure that the COMP and PROCESS LEDs are on and you will be able to hear the full compressor simulation. You can freely alter the attack, release, threshold and slope. In addition you can vary the PROCESS DRIVE to alter how hard you are driving the analogue simulation of the compressor effect.

### **Saving the whole compressor program**

Once you have an effect that you like, save it as a program for future use. See "How do I Save the setup for the next session?".

## **Sample Effects "Off-Line"?**

It is possible to make a recording of the analysis tone from Replicator and take this to a remote location before playing the sound through a signal chain you want to sample. If this is recorded again you can bring it back to Replicator and then sample it using the techniques described in "How do I Sample a Vinyl or Acetate Disk?".

### **Applications**

1. Your FX8000 is busy in the studio but you are going to another session. Take a recording of the analysis tone and run it through any nice sounding signal paths to a recorder. Sample it later when you get back to your FX8000.
2. Send a tape to distant site and have them play it over the phone to you while you record it or sample it locally. You can then replicate the phone line should you need this effect.
3. As above but using a better quality landline.
4. As above but using an FM or AM radio channel (but avoid any heavy compression).

### **Things to Remember**

1. This works best if there is no compressor element in the sample chain. - this will distort the level balance between samples but you may still get a usable effect by trimming the "Level low-high" down in the edit screen.
2. The whole chain is sampled from Replicator back to Replicator so use the highest quality recording medium possibly - unless this is what you are trying to sample!
3. If the timing is sample rate accurate throughout you will get the best results. Record the analysis signal using a DAW sample rate locked to Replicator, then do the remote sample out and back to the same DAW. Final analysis should then be sample rate locked which will eliminate the need for de-jitter.
4. If sample rate locking is not possible, use the Replicator "De-jitter" option in the sample edit screen. This is a very effective fix when sample rate lock is not possible.

## **Sample a signal processing chain?**

This enables you to (a) keep the setting for future identical use, (b) free up the equaliser for another use, or very importantly (c) apply the setting of a mono equaliser identically to a stereo or multi-channel signal. This means that you do not have to tie up your invaluable pair of classic equalisers for an EQ across a whole mix when you really need them for individual channel. (And you probably couldn't set them identically anyway!).

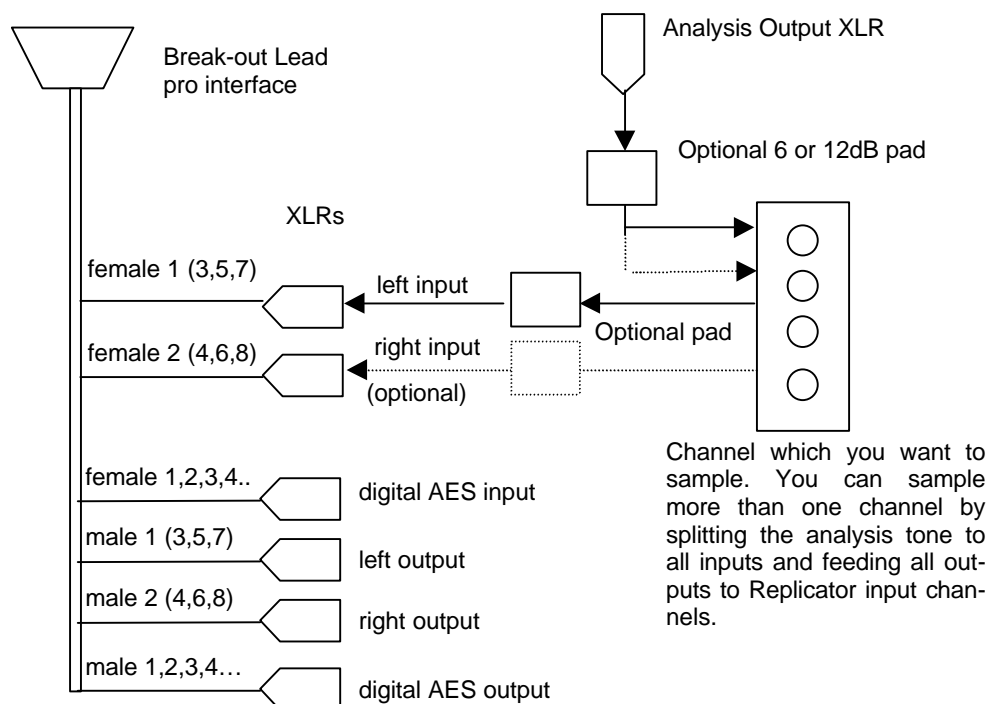
### **Does The Chain Contain a Compressor?**

If the processing chain you are sampling contains a compressor (or is a compressor) you first need to sample the sound of the chain with the gain reduction fixed at 0dB. Set the threshold to maximum on the compressor and also any time constants to maximum and any ratio to 1:1 if available. During the sampling tests take care that you do not get any gain reduction showing. If you do, and the thresholds are set as high as possible, we recommend reducing the sampling signal level to prevent this happening on peak sampling impulses.

After taking the signal path sample, restore the compressor settings and see "How do I sample compressor curves".

Note that if there are any other gain altering devices, for example noise gates, these should be switched off.

## Connecting Things



The diagram shows a simple setup for sampling an analogue channel. You probably already have the main left and right outputs connected to your monitoring setup and may have digital inputs and outputs connected.

If you are working at 48kHz or below you may alternatively be monitoring on the mix output (RCA phono jacks, consumer levels) or the SPDIF outputs (coaxial or optical).

The Replicator sample signal output is produced from the Analysis Out XLR. This signal must be connected to the device you want to snapshot. The same signal is available on the 3 pole jack attenuated 20dB which you might use for connecting to instrument amplifiers, etc. You could also use this if hum loops are a problem by using the two active signals to the instrument amp and not connecting the jack ground.

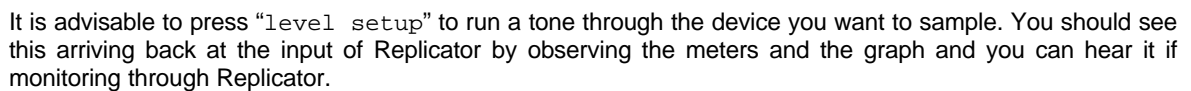
The output of the channel to be sampled must be patched into the analogue input of the Replicator as shown. You can use your own Analogue to Digital converter and feed into a Digital Input of Replicator if you prefer. Remember to select the "balanced" Input or the "AES Digital" input on the Input/Output Screen of Replicator depending on the connection. You can also use the consumer level input (RCA pin jack) if appropriate and select "unbalanced" input.

## Sampling Things

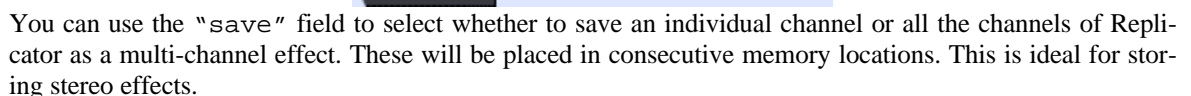
You get to the main sampling screen by selecting the system page (press SYSTEM), then selecting "set-up" (the top SELECT button), then select "sample" (the middle SELECT button).

You see this screen:

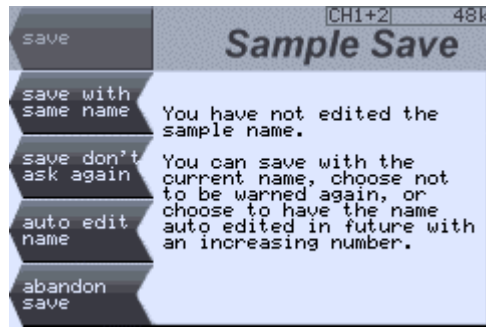




- Finally, this screen will appear:



- If you have not edited the sample name you will see the following screen...



The options give you the following choices:

5. Do the save with the unedited name
6. Do the save with the unedited name and do not present this screen again (until reset by the next power down)
7. Append a 3 digit number automatically in the last 3 characters of the sample name. This is incremented on every future save and you will not be prompted with the query again (until reset by the next power down)
8. Abandon the save - this returns to the save screen and allows you to manually edit the name or abandon doing it altogether.

The automatic numbering is useful when you are saving a large number of consecutive samples in one session .

## Listening to the New Effect

You should now see "How do I Listen to a Sample?" to audition the new effect.

## Save the setup for the next session?

Save the current setup as a PROGRAM.

Go to the program save screen... SYSTEM -> programs -> save

Select an empty or unwanted memory location, give a suitable name for the program and press "do save" .

The setup can be recalled at any time by loading the same program.

If you want to take the set-up away with you, you should upload the program *and any associated samples* using USB or MIDI to a PC for archiving.

Remember that the samples you used must be uploaded and stored as well as the program.

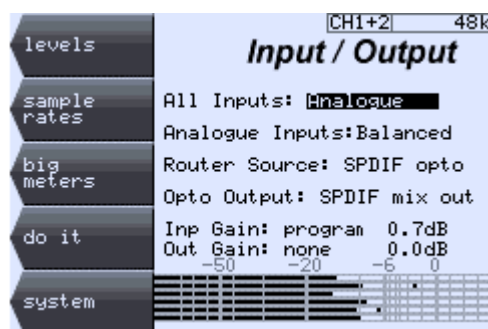
If you were using a Classic EQ effect, this cannot be uploaded. It could be

- Part of the standard Replicator library which may be obtained from the Sintefex Audio web site, or
- A special effect created for a Replicator customer - you would need to obtain a copy from the owner of the Replicator to be able to reproduce it.

## Select an Input?

Select the group of channels you want to select the input for. (Input can be selected differently for each group). Do this in the SYSTEM page (if you have a stereo group on a 2-channel system or a group that occupies all the channels of a multi-channel system this has no effect of course).

Press INPUT.



You can select the input source for all channels at once or for the current group of channels you are working on. Select "All Inputs" or "This Input".

Select the channel inputs from the list, which includes "Analogue" and "Digital" inputs. If you are operating at 48kHz or below you can also choose "Router", which selects inputs from the master section including SPDIF and ADAT.

Suitable equipped FX8000 systems also offer "Analogue DC" as an input option. This is the same as "Analogue" except that the digital high pass filter present in the Analogue to Digital converter is switched off. This is usually inaudible and should not be necessary but some engineers prefer this option. The downside is that you get some DC offset into the digital signal. The upside is that the filter which is down 0.1dB at 6.5Hz (13Hz at 96k sampling) is removed from the system. Your ears can decide. However all channels must be selected "Analogue DC" for this to be effective - you cannot mix and match across channels.

If you are using Analogue inputs, you can select between "Balanced" and "Unbalanced".

- Balance and Unbalanced analogue input apply to the whole machine, not individual channels.

Balanced inputs are available on the multi-way connector at professional operating levels. Unbalanced inputs are available on RCA phono-jacks at commercial operating levels.

If you are using a mixture of analogue and digital inputs you need to make sure you have the system synced correctly. See "How do I Sync to master sync?" for more details.

### Using the "Router" at 48kHz or below

The router source may be selected from the list of "Instrument", "SPDIF coax", "SPDIF opto" and "ADAT".

The Optical Output may also be used at 48kHz and below for output of SPDIF mixed signal (same signal as on mix analogue output) or for output in ADAT format, either of the mixed signal to each pair of channels or of each Replicator channel to the matching ADAT channel.

## Speed up the MIDI or USB?

Every Replicator sample is about a Megabyte in size, and every program is 32k bytes. Because a program often uses more than one sample you can see why floppy disk is no use for transferring effects around.

Classic EQ effects can be typically up to about 6Mbytes.

MIDI is limited in speed to the MIDI standard of 31.25kbaud, which is about 3kbytes per second, so a sample will take around 300 seconds (5 minutes) to transfer on MIDI, and an effect up to an hour.

USB can be a lot faster - up to 100 times faster than MIDI. This is what you get with the "standard" USB setup (see the remote control page).

## Switch a section in and out for comparing the effect?

Press the CALL/BYPASS button for a processing section.

Immediately the page for that effect is called (hence the name "CALL"). If the button is held down for more than about half a second the LED will start to flash. When the button is released the section will toggle, i.e. turn off if it was on or on if it was off.

## Switch the whole system in and out (bypass) for comparing the effect?

The SYSTEM button works differently from the individual section bypass buttons. It instantly bypasses the whole system including input and output level controls. The bypass is "digital" however. Analogue signals still pass through the A-D and D-A converters.

## Switch off Replicator if the POWER button does not seem to work?

Because the POWER button is under the control of the software (to prevent accidental switch off), it is possible for the POWER button to be inoperative if there has been a software problem, or if the software has not fully loaded. In this case power should be removed for a minimum of 10 seconds.

When you re-power the system, the POWER switch should be operating again.

The POWER button does not operate if you are

- Running the Engineering Software "Eng App", or
- Running the START ROM,
- In the middle of a long software operation that should not be interrupted,
- Downloading digital audio data dump files to the SPDIF input.

## Sync to master sync?

Select the Sample Rate page:

SYSTEM -> clock

The master clock source can be selected from the list.

- automatic selects the reference to the digital input if you have selected a digital input
- internal uses the in-built clock generator and is used for analogue work or if Replicator is master in a digital system
- AES ref selects the AES ref input as source of digital clock
- AES 1+2 selects AES input on channel 1 and 2 as the reference
- SPDIF selects the SPDIF as the reference
- wordclock selects the BNC wordclock input
- ADAT selects the ADAT input as a reference.

Replicator cannot sync to digital inputs 3 to 8. If you feed digital to these channels it must be in sync with a digital input to channels 1 and 2 or to AES ref, either of which may be used as the source of reference clock. Automatic locking will select locking to AES input 1 and 2 if any channel group input is set to digital.

## Use Delay Compensation?

Delay Compensation allows the FX8000 system delay to be set according to a number of options.

- Minimum Delay: This causes FX8000 to operate with minimum processing delay at all times (this is the way the FX8000 has worked until now).
- Bypass Matched Delay: This causes the BYPASS setting to match the processing delay when the system is in.
- Fixed Delay: This fixes the FX8000 processing delay at 150 samples, regardless of the effects selected (except for POWER LINEAR (linear phase) EQ).

To get to the delay options select...

SYSTEM -> setup & sample



A different mode may be used on each channel if working in mono or on each group of channels in a multi-channel system.

- Some effects may exceed the specified delay or not be matched in BYPASS, for example linear phase EQ or reversed samples. The delay is displayed in the main system page in mS as measured from analogue in to analogue out.

### What are the advantages of Delay Compensation

The minimum delay is often the best choice for real time work, for example recording or live use. In addition selecting the BYPASS to processed signal delay can make comparing the effect with the bypass easier as the timing relationships remain unchanged.

The fixed delay mode allows multiple channels to be used with a consistent delay.. This delay, although small, may be critical in a real-time situation or a multi-track mix. A change in delay when the system is switched to bypass may change the acoustic effect of an instrument in a mix, especially when it may be one of many microphone channels in a setup, for example a multi-miked drum kit or orchestral performance.

### Some points to watch:

- It is not possible to match the delay of Linear Phase EQ as this typically amounts to about 2000 samples.
- Samples effects incorporate delays dependent on the device sampled, so delay compensation is only as accurate as the original sample. Samples are assumed to have peak response at 20 sample delay unless a delay is specified in the sample file. In addition some delays are frequency dependent so delay matching has to be a compromise.
- The processing delays are in addition to input/output delay. The Analogue In to Analogue Out has a fixed overhead of about 1.3 mS (about 63 samples) at 48kHz. Digital In to Digital Out is shorter.
- AFX delays are not compensated as they are considered intentional additional delays. The correct delay is calculated for the dry component in the wet/dry mix signal in the AFX section. If the AFX loop is set up to include sampled reflections (for example) this becomes part of the "wet" signal and is not compensated.

### Use Delay Effects (in the AFX section)?

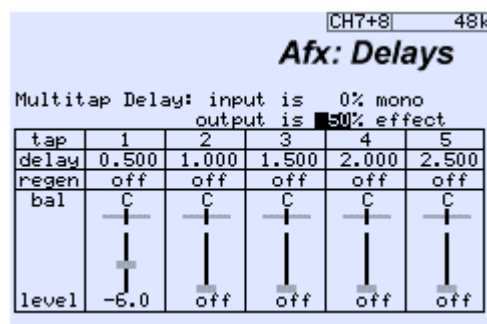
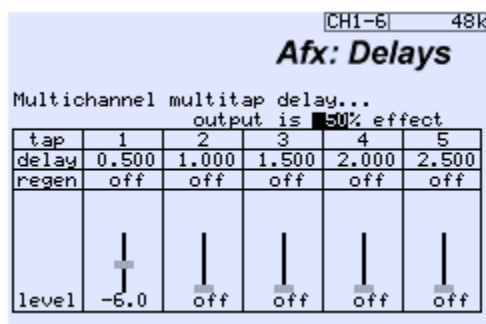
The AFX section offers a multi-tap digital delay line at both 44.1/48kHz and 88.2/96kHz sampling rates. At the higher sample rates 16 seconds per channel of delay is available, and at the lower rates 32 seconds is available.

5 taps are available from the delay line each with delay variable over the full range in 1mS steps.

These delays can be used for musical effects, mono, stereo or surround sound matching delays for sound to picture or live use, profanity delays in stereo or surround productions, or as a cutting delay in lacquer disk cutting.

This section is selected using the AFX button. Press this and select "multidelay" from the "After Effects" screen.

If you are working in mono or surround you will see a screen like the one on the left here:



This shows that each of the 5 taps can be faded in and out and mixed together. The field "50% effect" can be altered from 0% (no effect) to 100% (all delays and no input signal). The delay of each tap can be set in 1ms increments between 1mS and 32-seconds (16 seconds above 48k sampling).

There is also a regeneration control for sustained echoes for each tap.

If you are working with a stereo group there are some additional controls, as shown in the picture on the right.

Here the feed to the delay line can be pre-mixed to mono, or partially to mono. Use the field "input is 0% mono" to vary this mix. 0% mono means the stereo channels are not mixed, while 100% mono means they are fully mixed.

When the stereo feed is mixed to mono you will often want to pan the outputs of the delay line, and the balance control can be used for this.

- This is a true balance control for the stereo delay line output tap so when the input is fully stereo this balances the left and right channels.
- The balance control varies between -10 for fully left and +10 for fully right. In addition it can go beyond this, which will introduce antiphase elements into the opposing channel. For instance selecting the control to +15 will give you maximum level on the right in normal phase, with the left channel reduced 6dB in antiphase.

These antiphase effects can be useful in musical mixes although, of course, the usual care must be taken that the effect is mono compatible if this is an issue for the particular application.

- At the higher sampling rates the AFX multitap delay is not always available depending on the other effects in the processing chain on the channel or channel in use. This is not usually a problem as delay effects are often used independently when other Replicator functions are not needed. Usually at least one of the other Replicator effects may be used, digital compressor, Classic EQ, digital EQ or snapshot signal path simulation.

## Use Different Samples on Different Channels?

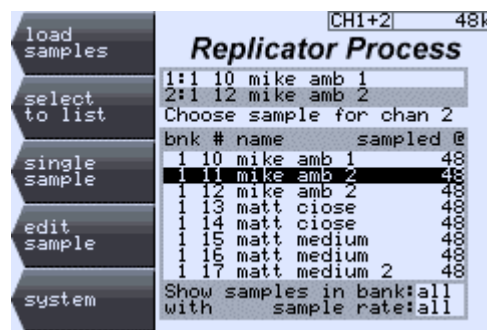
There are two ways to use different samples on different channels.

If the channels are to be used completely independently, set them up as separate mono channels. See "How do I Use Two Mono Channels on a stereo machine, or channel combinations on a multi-channel machine?". Then you can use completely independent settings. (Mono compressors can still be stereo linked in pairs).

If you want to build a stereo of multi-channel effect with different samples on each channel - for example a stereo pair of effects to be applied to a mono source, you can do this as follows:

1. Set up the channels into a group, (see the reference just mentioned above)
2. Select a sample for the group (see "How do I Listen to a Sample?"),
3. Select PROCESS -> select samples -> multi-sample.

You will see the following page:



There are two panes to this window. Select a sample for one of the channels in the lower pane, then move the cursor to the upper pane and select a channel to place the sample in. The press "select to list".

When you have built up a selection of samples, one per channel, press "load samples" and the samples will load into each channel as specified.

You can use "single sample" to revert back to using the same sample for all the channels of the group.

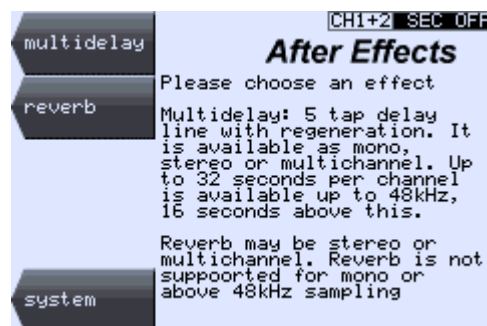
## Use Reverberation?

The AFX section offers a set of reverberation programs as an alternative to the AFX multi-delay option.

- Reverberation programs are available at sample rates up to 48kHz only. They are not available in MONO settings, but are provided for use in stereo (a different effect may be used on every stereo pair of a multi channel machine) and are also available in multi-channel configurations.

## Selecting the Reverberation Effect

Press the AFX button from any program where an AFX multidelay has not already been selected. You will see the following screen.



Press the "reverb" SELECT button to go to the reverb screen.

- If you have an AFX multidelay already selected, you can switch to reverberation by pressing the "Change to Reverb" SELECT button.

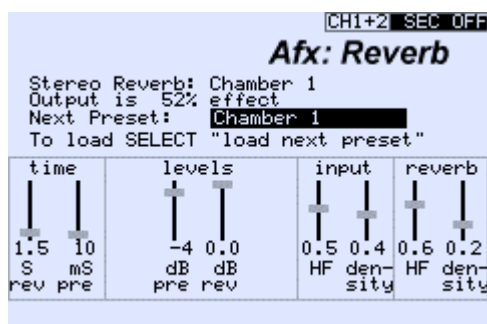
The reverb page appears and if AFX is switched on (by holding the AFX key for about half a second) the default reverberation can be heard.

To select a particular preset, scroll through the "preset:" options using the DATA knob. Then bring up the SELECT options by touching any SELECT key. You will see a screen like this



Press "select preset" to load the new preset.

You will see a screen like this, which we will now explain...



- You may see up to four faders in the "levels" section. The options available depend on the reverb program and other factors, described below.

## Summary of Reverberation Programs

Small Plate	Bright plate better for shorter reverberation times
Medium Plate	Larger plate better for medium to long reverberation times
Room 1	Ambience of a small room, opens up with longer reverberation times
Room 2	A different sounding room
Chamber 1	A thicker reverberation chamber
Chamber 2	Larger chamber
Hall 1	Hall with significant build up of reverb field over the first second or so
Hall 2	A much larger hall giving "cathedral" like delay in reverberation field build-up, For long reverberations only.
Non-lin	Reverberation does not decay exponentially but stops suddenly, as if gated. Duration can be adjusted to match tempo.
Reverse	Like Non-lin but the reverberation field builds up to a crescendo at which point it stops suddenly, giving a reversed reverberation effect.

## Wet/Dry mix

A mix between the reverb effect and the dry signal is shown at the top of the screen where it says "Output is 50% effect". This may be varied, and you will usually set this to 100% effect when you are using the reverberation in the echo send loop of a mixing system. The middle select button can switch you directly to 100% effect.

## Reverb time and pre-delay

The section labelled "time" allows control of reverberation time up to 20 seconds also allows pre-delay to be added to the reverberation up to 999mS.

Reverberation time is an approximation to the standard RT60 definition but since this measurement depends on the amount of reverb mixed into a final signal the value is only approximate.

In the case of "non-linear" reverb programs the reverb time display is replaced by a simple length variation control which allows the period of the effect to be adjusted easily to match the program material.



## Mixing early, middle and late reverb signals

All reverb programs have the "reverb" level fader available.

A reverb program may also have early reflection signals ("pre") and late reverb signals ("late"). If these are available for a given preset the faders will be shown. Early reflections represent clean delays associated with early reflection from hard surfaces in the artificial reverberation space. Late reverberations are additional reverberation "taps" placed beyond the normal reverberation space and can be used to thicken and extend the reverberation effect if desired.

The fader labelled "sim" allows mixing in of sampled early reflections and is used when a reverberation program is used together with a Replicated sample of early reflections in a simulated space. This mixes in the sampled early reflections to the combined reverberation signal. To set up sampled early reflections, see "adding sampled early reflections" below.

## Input Section

The signal entering the reverberation space may be pre-conditioned by attenuating the HF content using the HF fader. Real spaces usually have high frequency absorption so unless you are trying to simulate very hard surfaces you will usually need some HF attenuation.

In addition you can use the density control to diffuse the signal before it enters the main reverberation space to break up very percussive sounds. This prevents many very discrete echoes of percussive sounds. You may need less of this if you are reverberating non-percussive material or for special effect.

## Reverb section

Once the signal is in the reverberation space its decay depends on the HF control in this section. Lowering the HF here causes the higher frequencies to die away faster than lower frequencies as it would do in real spaces.

The density control varies the way the signal is further diffused as it reverberates. A low value here would simulate many parallel surfaces where discrete echoes can be discerned. A higher value would diffuse the reflection into a general field of reverberation.

## Non Linear programs

Loading a non-linear program introduces a reverberation effect where the effect does not decay normally, but cuts off after a certain period. The "time" section lets you adjust the duration using the "non lin" fader.

The "Non-lin" and "Reverse" programs are two examples of non-linear reverberation.

## Using EQ or Compression with Reverb

The Reverberation processing of the FX8000 uses a large proportion of the available resources which introduces some limitations into what else the FX8000 can do at the same time.

However it is possible to use any one other process of the FX8000 with reverberation. In this way you can apply compression, digital EQ or Classic EQ to the reverb signal either before or after the reverberation process.

Sampled early reflections can also be applied where you have ambient room samples to generate interesting new reverberation effects. See the heading "Adding sampled early reflections".

Classic Compression is not available (although ordinary digital compression is) as this requires both the Replicator PROCESS section and the compressor section to operate at once.

Reverberation is also unavailable at sampling rates above 48kHz.

## Adding sampled early reflections

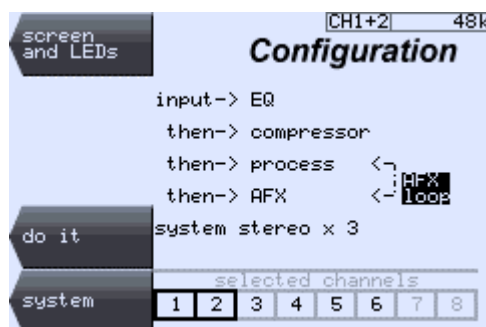
If you have a sample of a real room with early reflections made using the Replicator FX8000, you can load this sample into the PROCESS section of Replicator in advance of the AFX reverberation. You can then use

## Sintefex FX8000 Analogue Sampling Digital EQ / Compressor / Effects Unit

the wet/dry mix to generate a combined effect of sampled early reflections and any reverberation program that can be mixed externally.

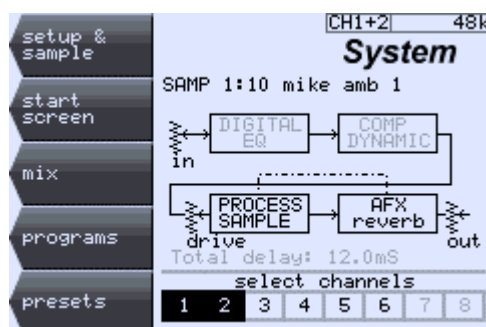
If you wish to balance the sampled early reflections with the reverberation field and still have a wet/dry mix available within the FX8000, you can tie the sampled effect into the AFX reverberation by means of an AFX loop. This enables AFX to "include" other FX8000 sections within its wet/dry control.

Select the SYSTEM->setup & sample->config screen as shown here...



You can move the cursor to the "AFX loop" area, then the DATA knob allows you include any previous sections into the AFX system. Normally this would be the PROCESS section of the FX8000 as shown here. Press "do it" to make it take effect.

This will be illustrated in the SYSTEM screen as shown here, with a dotted line linking AFX to the included effect.



When you have selected this option, the "through" level fader will be available in the main reverb screen to allow mixing of the sampled early reflections with the reverberation field.

- The AFX loop is also used in the "multidelay" program to feed back the signal in the echo loop to an earlier section of the FX8000. Thus an echo can be placed round a tape simulation for example to simulate more accurately tape signal degradation with repeated passes round the loop. EQ or compression can also be included in the loop. EQ will allow progressive change to the signal and compression can be used to maintain constant echoes which neither amplify nor fade away.

## Surround Sound or Multi-channel reverb

Multi-channel FX8000 units can be used for 4-channel, 5.1, 6.1 or 7.1 surround reverberation, or for multiple independent stereo reverbs. Every stereo pair can have a different reverb program.

An 8-channel unit can also do two 4-channel reverbs, a 5.1 (or 6 channel) reverb and a stereo reverb, etc.

In general the channels have been programmed with the following channel arrangement in mind. However, you may want to experiment by patching different Replicator output channels to different mix channels, as each output channel, although correlated, represents different acoustic taps into the reverberation space. For all surround effects it is not recommended to process the LFE channel. For all reverb modes the all channel inputs are summed into the reverb feed.

Channel	Spatial Location
1	front left

2	front right
3	rear left
4	rear right
5	front centre
6	rear centre
7	wide front left
8	wide front right

*Typical Channel Patch for Multichannel Reverb Programs*

## 4-channel surround

The 4 inputs are summed to feed the reverberation processor, which produces a diffuse 4 channel output with channels 1 and 2 intended for the front stereo pair, and channels 3 and 4 for the rear stereo pair. The dry/wet mix within the unit allows the 4-channel mix to be blended with the reverberation field.

## 5.1 Surround

5.1 reverberation sums all six inputs to feed the reverb unit. In general the Low Frequency Effects (LFE) channel should not be processed through Replicator. If it is passed through it will participate in the effect.

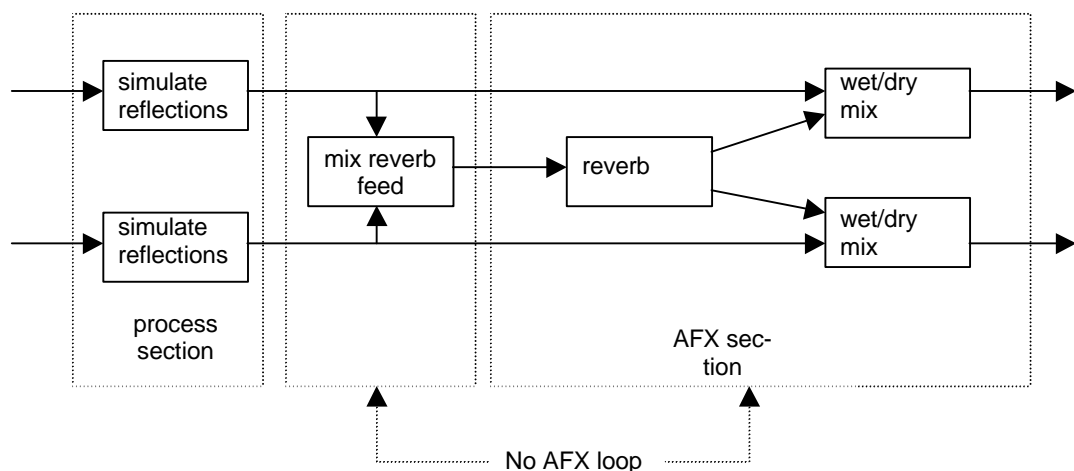
## 6.1 Surround

This can be produced on a 6 channels system as it is not usually necessary for the LFE channel to participate in reverberation effects. The rear centre is typically derived from channel 6.

## 7.1 Surround

7.1 operation sums all 8-channels as the reverberation source and provides two additional reverberation feeds on channels 7 and 8 intended for a widely separated front pair as extreme L and R.

## Signal Path when AFX loop is not used, showing optional early reflections



The stereo signal is mixed into the reverb path immediately before the reverb, which then may be mixed back with the stereo signal within the FX8000 if a desired for wet/dry mix.

Multi-channel operation is the same as stereo - all the channel get summed into the reverb feed.

The diagram illustrates the AFX algorithm architecture, organized into three main sections: **process section**, **AFX section**, and **AFX loop**.

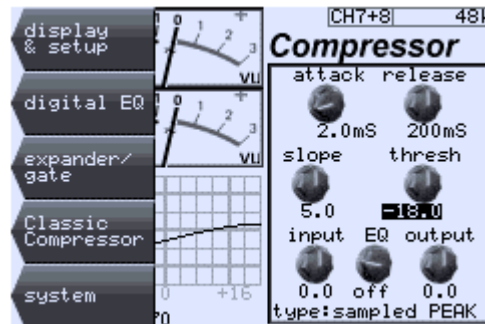
- process section:** Contains a **mix reverb feed** block. It receives input from the **AFX loop** and feeds into two **simulate reflections** blocks.
- AFX section:** Contains a **reverb** block and two **early mix** blocks. The **simulate reflections** blocks feed into the **early mix** blocks. The **reverb** block also feeds into the **early mix** blocks.
- AFX loop:** Contains two **wet/dry mix** blocks. The **early mix** blocks feed into these **wet/dry mix** blocks. The output of the **wet/dry mix** blocks feeds back into the **AFX loop**, which then feeds into the **mix reverb feed** block.

For those who wish to experiment, the reflection simulation can actually be any sampled effect. It can also be a Classic EQ, of even a compressor or digital EQ.

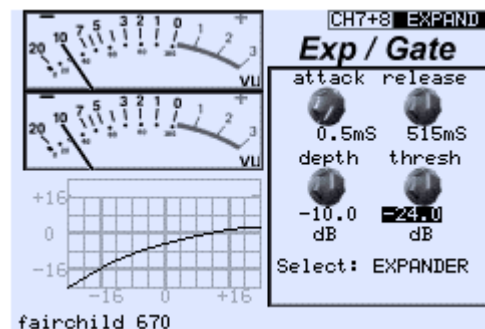
If you use the cursor keys to get to a character in a string and before the CURSOR key is released, you turn the data knob clockwise, the right hand part of the legend can be scrolled to the right to make room for inserting text or to clear the field.

## Use the Expander/Gate?

Select the COMPRESSOR VIEW button, then touch the OPTION SELECT buttons to get the options shown here.



Touch the "expander/gate" OPTION to select the Exp/Gate page shown here.



You can adjust the various parameters and use the "Select:" field to choose between "OFF", "EXPANDER" and "GATE". When either channel is actually expanding the status area will show "EXPAND" as shown here. In addition the bottom LED in the gain reduction window also lights to signal the expander gate is in operation for each channel that is affected.

Attack is the time it takes the gain to recover when a signal re-appears after expansion or gating is in operation. Release sets the rate at which the gain reduces when the signal falls below the threshold.

The expansion curve is 2:1 and the maximum gain reduction that the expander/gate gives is set by the "depth" control. This gives an easy control of expansion.

The expander and the gate differ in that if gating is in operation, the threshold automatically increases by the amount the gain reduces. This allows a clean "switch on" or gate opening when the signal exceeds the higher threshold.

The expansion and gate characteristics are not sampled from original analogue equipment, but are provided for use with the sampled compressors to add to the functionality of the original device. This allows you to recover dynamic range lost in compression while preserving the sound of analogue compressors (if a Classic Compressor is loaded) or to augment the Classic EQ if you are using that.

The gain reduction shown is the combined gain reduction of the compressor and the expander gate. The compression curve shown is for the compressor section, even when you are looking at the expander/gate page.

You can return to the main compressor page by choosing the OPTION "back" or by pressing the COMPRESSOR VIEW button again.

- The expansion or gate gain reduction is never linked to other channels. Each channel operates independently.

## Use Two Mono Channels on a stereo machine, or channel combinations on a multi-channel machine?

A two channel machine may be used as a stereo processor or as two separate mono processors.

A multi-channel machine may be used for various combinations of multi-channel, stereo and mono processing.

The types of grouping available are

- 7.1 surround (8 channels acting together).
- 5.1 surround (6 channels acting together, also used for 6.1 surround with 6 channels acting together, with LFE not passing through Replicator).
- 4.0 surround (4 channels acting together).
- stereo
- mono

If you have enough physical channels, you can select combinations of these according to the following rules:

The grouping with the most channels comes first (starting at channel 1)

The grouping with the same or the next fewer number of channels comes next, (and so on until..)

Any remaining channels can be set up as mono channels.

You select the configuration using the Configuration page:

SYSTEM -> setup & sample -> config

You will see a field labelled "system". Select this and you can scroll through the combinations available. You need to press "do it" when you have selected a configuration you like.

The cursor can be placed in the channel boxes at the bottom of screen and the DATA wheel can select between the channel groups. Whichever is selected in the one the front panel controls will be affecting.

## View the VU or bar meters?

For viewing input levels select

INPUT -> big meters

Touch any SELECT key and the vanishing labels appear which allow you

- setup the levels (especially for VU meters)
- toggle VU and bar meters
- view all the channels or just the current group (you will see no change if the current groups is using all the channels fitted)
- return to the system page

These vanishing labels will disappear if you do not use them for 15 seconds or if you touch the DATA wheel or CURSOR key.

## 4 General Information

### Warning and Error Messages

These are some of the warning and error messages you may get, what they mean and what to do about them. Some of these require a confirmation before proceeding while some are reported for information without stopping operation.

<b>message</b>	<b>means</b>	<b>action</b>
Cannot select SPDIF, it is in use for compressor external gain link.	You are trying to select SPDIF as the input or output format, but it has already been selected for compressor external gain linkage.	Go to the compressor setup page and select the external gain linkage to be "off". The SPDIF connection cannot be used for audio at the same time as it is being used for gain linkage.
Do you want to save with the unedited name?	You are about to save a program but have not edited the name.	Either continue to save with the old name or go back, edit the name and then save again. Too many programs on disk with the same name can be confusing.
Effect not found	The program is loading a classic EQ or compressor but the requested effect is missing.	You should load the missing effect from the latest library release, and use the "tidy disk" feature to sort the effects into numerical order.
For full reset, power off for 10 seconds	A short power interruption has caused a system error.	Power down for at least 10 seconds before re-applying power. You may need to use the rear panel switch (if fitted) to interrupt the power. Consider conditioning the line power with an uninterruptable power supply in areas where power supply is unreliable.
graphic -nn -or- no bitmap -nn (in pre v2.1 software)	System software has been upgraded but the matching graphics have not been updated. nn is the internal identifier for the missing graphic.	Complete the upgrade by running the latest version of Replimat (at least v2.1) and reloading the software.. The system will work without it but the display will be wrong.
Memory already in use, do you want to overwrite it?	You are trying to save a program over an existing program.	You can abandon it or overwrite the current contents of the program memory.
Memory empty, using unit impulse	A program tried to load a sample that is not stored on the disk.	Load the latest sample banks or the specific sample required by the program. Check that the program is the latest available if a library pre-set.
No program is selected, check search bank and sample rate	You tried to load a program when no valid program was selected in the listing box	Usually this means you are looking in an empty program bank. Scroll to "Show programs in bank n" and select a different bank or "all".
Please register this software - it is quick and easy to do...	The software is reminding you that it should be registered for continued use.	There is still plenty of time for evaluation so you can ignore this. If you have purchased the system please register the software for unlimited use.
Program's sample rate differs, load anyway?	You have selected a program saved at a different sampling rate.	You can try loading it but it may not be appropriate for the current sampling rate. Often it is better to change sample rate and try again.
Registration of this software is free to all end users...	The software must be registered with Sintefex Audio for use beyond a period of about 1000 hours of use. This is free of charge to end user custom-	The evaluation period can also be renewed by distributors or system evaluators. See "How do I Register This Software?" for more information.

## Sintefex FX8000 Analogue Sampling Digital EQ / Compressor / Effects Unit

<b>message</b>	<b>means</b>	<b>action</b>
This selects an unlocked digital source. Check the input signal.	You have selected a digital input or clock source that is currently not locked to a valid input.	You should connect a valid input (at which point the message box will go away and the operation will complete). You can abandon the change or go ahead anyway if you know the signal will be connected later.
This software must be registered for continued use...	You must register the software with Sintefex Audio for uninterrupted use of the system. All users can do this free of charge. Evaluation users and distributors can request a renewed evaluation period.	End users should make a note of the code on the "register software" page and send this to Sintefex Audio using the card in the manual for a permanent authorisation code. When this is entered this message will no longer occur. Evaluators should contact Sintefex Audio for a renewal authorisation. See "How do I Register This Software?" for more information.
USB was suspended because remote computer was shut down.	Replicator was attached by USB to a computer (host) or hub that was powered down. To Replicator this is indistinguishable from continual reset requests from the host. Accordingly Replicator has temporarily suspended the connection.	This message is shown in the "Remote Control" page on Replicator which will continue to operate normally from its front panel. To re-establish USB connection, ensure the PC host or USB hub is powered up, and that "USB setup: standard" and "USB Remote Control? yes" is selected, and press "do it".
Warning, matching effect has changed since program created	The program is loading a classic EQ or compressor but the requested file is different from that expected.	Either the program or the classic effect is out of date. You should reload the affected program and effect from the latest library release.
Warning, matching sample has changed since program created	The program is trying to load an FX8000 sample but the requested file is different from that expected. No sample is loaded.	Either the program or sample is out of date. You should reload the affected program and sample from the latest library release.
Warning: You need at least software version n.n.n for this Classic EQ/Compressor	You are trying to load a Classic EQ or Compressor that requires a more up to date software version than your Replicator currently has installed. You cannot load it as the software does not know how to handle the new data.	Update your Replicator software to the latest version. This also ensures you have the latest features and bug fixes. If n.n.n is for example 2.2.0 this indicates that software 3.0.0 and 2.2.0 would be okay but 2.1.2 and 2.0.0 would not be okay. Usually new software releases just quote the first 2 digits, e.g. 2.1, 3.0 etc.

## Status Display Messages

At the top right of the LCD display is an area that shows various messages ranked in order of importance. The following table shows a summary in order of importance (i.e. the higher messages in the table will appear instead of the lower ones if the warning condition applies). Warnings above BYPASS (and external reference that is more than 5% out) will also flash to indicate a condition that requires attention.

<b>message</b>	<b>means</b>	<b>action</b>
no ref	external digital reference is selected but no reference is supplied.	Check the digital input has a good signal if audio input is digital. Check that the clock reference is selected to automatic or if not that the input selected is fed a valid digital signal. Try a different connecting lead. If all else fails use an internal reference.
unlocked	The system has not been able to lock to the reference input even though it appears to be present. It may be at a sample rate not supported by FX8000 or it may be too unstable to lock to	Check the quality and sample rate of the reference signal. Try a different connecting lead. If all else fails use an internal reference.
IN CLIP	The current input is producing peak digital signals (clips). This may be acceptable if your ears tell you that the clipping is not a	If it is an analogue input you will need to reduce the signal level into the FX8000. If it is a digital input you need to reduce the output of the device sending digital signals. If this is a DAW it may be set to hit digital



<b>message</b>	<b>means</b>	<b>action</b>
	problem, but be aware that adding EQ (for example HF lift) can make these clips noticeable later in the signal processing.	peak from time to time, in which case you can ignore the warning. If you are habitually getting overload from analogue signals you can trim the input gain of the FX8000 by fitting a gain adjustment header. See the installation information.
MIDI OVF	MIDI messages arrived too fast to handle, some have been discarded	Slow down the rate at which your sequencer is sending parameter update or program load messages to the FX8000.
BYPASS	The system is bypassed on the channel(s) you have selected.	Press "SYSTEM" button in the CALL / BYPASS section to enable the FX8000 to process the audio.
SEC OFF	The section (EQ, compressor) you have displayed on the LCD is bypassed.	Touch the "SECTION" button on the EQ or compressor to enable the section. If it is the EQ section you may need to load a Classic EQ if one is not currently loaded. Press the EQUALISER VIEW button to get this option.
exp1, exp2	The Expander/Gate section of the Compressor is reducing the gain on the channel indicated to expand the dynamic range.	If this is not what you want, select the COMPRESSOR VIEW and touch the OPTION buttons to select the "expander/gate" option. Select the Expander/Gate to "OFF" or adjust the controls to taste.
Gate1, Gate2	The Expander/Gate section of the Compressor is reducing the gain on the channel indicated by gating the signal.	As above.
ext 48k, 96k etc	This shows the sample rate currently in operation, locked to an external reference. If this is on a BLACK background it warns you that the sample rate is more than 5% away from the selected sample rate.	If the message is on a BLACK background, you may have the wrong sample rate selected in the FX8000 or it may be receiving the wrong sample rate reference. This may be okay if you are (for example) using varispeed. If it is unexpected, select the correct reference clock and the correct sample rate for the FX8000 using the INPUT->clock option.
48k, 96k etc	This shows the current internally selected sample rate.	You can change this using the INPUT->clock option if this is not what you wanted.

## File Types

Users who are using a PC for system management and update will be creating a number of files on their PCs. This is a summary of the file types and the extension (characters after the dot) they use.

<b>extension</b>	<b>file type</b>	<b>more detail</b>
.rpg	Replicator Program	Settings of all controls on Replicator are a program. When stored on a PC they have this format.
.rsm	Replicator Sample	FX8000 users can create samples of signal paths, and the FX8000 library contains some of these. They can also be used on FX2000.
.rfx	Replicator Effect	This file contains a classic EQ or Classic Compressor and is supplied by Sintefex Audio for download to Replicator machines to add to the library of these samples available.
.pdf	Acrobat document	A document that can be read on any computer with an Acrobat reader, available free of charge from Adobe. This is the main method of distributing documents on the internet as it is independent of computer type and word processor package.
.dll	A part of Windows software applications.	Windows applications consist of an .exe main file and a number of .dll files that perform part of the work. All must be present for correct operation.
.exe	The main part of a Windows software application.	See above. Replimat.exe is the main part of the Replimat program.
.sys	A driver file.	This is required to allow Windows to talk to external hardware, e.g. Sintefex Audio hardware
.ldr	A loader file of new FX2000 / FX8000 firmware	This is copied to the hardware to update the on-board firmware program. This allows new operating features to be

<b>extension</b>	<b>file type</b>	<b>more detail</b>
.inf	An information file.	This is used by Windows to manage installation of .sys files

## Glossary

Some of the terms used in this guide...

<b>term</b>	<b>means</b>
0dBFS	Zero dB relative to Full Scale digital. This refers to the peak signal level that the digital audio converters or interface can handle. A sine wave of this signal level will produce clip indications although will be passed undistorted. A sine wave at any higher level or any other waveform at this level will produce clip indications and will be distorted by the converters or the digital audio interconnection (AES or SPDIF). Internally the FX8000 can handle signals higher than this due to its use of 32-bit floating-point format. see "head-room".
24-bit	Digital audio can be to 16-bit accuracy as on a CD or to a higher accuracy as often done in master recording. 24-bit is the highest accuracy in use and can offer a dynamic range approaching 144dB from minimum dither noise to peak signal, and is never actually achieved in practice due to converter limitations. In addition every calculation must generate further inaccuracy. FX8000 does all processing on 32-bit floating-point audio data to preserve the best accuracy over multiple arithmetic operations.
AES	Audio Engineering Society - often used as a term for the professional digital audio interface standardised by the AES in 1992, in which a pair of signals is carried on a cable with XLR connectors. This signal can usually be sent a lot further and more reliably than the similar domestic SPDIF format.
Analogue DC	Input option in which the digital DC removal filter present in most analogue to digital converters can be switched off if preferred to extend low frequency performance at the expense of some digital DC offset.
Application	A program that runs on a computer under control of the operating system.
Classic Compressor	One or more samples of the signal processing path of a prized compressor together with one or more sets of gain reduction curve data which together contain all the data needed for recreating the sound of that compressor in a range of settings. The recreation is performed entirely digitally using the unique Dynamic Convolution process.
Classic EQ	A collection of samples of a prized equaliser, sampled in a large combinations of settings, and presented with an approximation of the control layout of the original equipment, and arranged so that settings equivalent to those available on the original equipment select the nearest sample or sample, often interpolating between them to get a very close approach to the original equaliser sound, reproduced entirely digitally using the unique Dynamic Convolution process.
Clock Rate	Same as sample rate, so called because the system sees the sampling as a precise clock tick at the sampling rate.
Compressor	An audio processor that reduces the dynamic range of programme material, turning down louder parts according to a chosen characteristic, often called the characteristic curve of the compressor. It is used for supplying audio to a less than perfect audio channel (for example AM radio) or for dramatic effect in music production.
Digital Gain Link	In order to avoid image shift in a stereo or multi-channel compressor it is necessary to link the gain so that the channel requiring the greater gain reduction at any instant determines the gain to be applied equally to all channels. In a digital system this gain linkage must be applied digitally and this is done within a unit or using a proprietary format between multiple units.
DSP	Digital Signal Processor - a computer chip optimised for processing the real time digital signal that represents an audio waveform.
Dynamic Convolution	A proprietary process for using a sample of an audio processing system or signal path taken at multiple levels to recreate the sound of that signal path digitally at a future time, including recreating the way that a signal processor can sound different according to how loud a signal is passed through it.
Effect	This is a general term for Classic EQ and Classic Compressor effects stored on an FX2000 or FX8000. Thus an machine may separately store Programs (that contain the settings of the machine), Effects (which contain the information about how to sound like an EQ or Compressor). In addition Samples can be separately stored on an FX8000
Expander	An audio processor that increases the dynamic range of programme material by turning down the level of quieter passages below a threshold. It is used for reducing low level noise and restoring dynamic range after compression.
Floating Point	In addition to the standard maximum audio sampling accuracy of 24-bit fixed point, Sin-

<b>term</b>	<b>means</b>
	tefex Audio processors preserve this accuracy and adds a further 8 bits of ranging data to create 32-bit floating point values. This means that the theoretical 144dB of signal to noise ratio is preserved through gain changing stages of compression and EQ right to the point where the 24-bits must be fed out again as accurately as possible. This gives about 800dB of internal dynamic range to the arithmetic processing.
Full/Pos/Neg	Full wave rectification for compressor level detector as opposed to half-wave where positive half cycle (Pos) or negative half cycle (Neg) may be selected. Half-wave level detection gives compressor artefacts rich in even harmonics, while full wave (which is more usual) can only generate odd harmonics. Even harmonics are often regarded as being more "musical".
Gate	Similar to an expander but with the additional effect of requiring a more significant audio event to restore the audio gain (open the gate) after the gain has reduced (closing of the gate). This is often used for dramatic effects and cleaning up percussion and other tracks of a multitrack recording.
Headroom	This refers to operating at signal levels somewhat below 0dBFS (or peak digital level) to allow room for unexpected audio events or for equalisation to take place without immediately encountering clipping. During the early days of CD manufacture it became normal to operate with almost no headroom due to early 16-bit systems sounding very unpleasant as the signal level was reduced. Modern conversion and 20 or 24-bit working should allow for operating levels some 12 or 18dB below peak as normal practice while exceeding CD performance and avoiding clipping. Final mastering to CD can then optimise the final levels to peak if this is desired.
Linear Phase	A method of processing audio in which all frequencies are delayed by the same amount. This preserves transient response and is often equated with a better stereo image. Total system delay is usually greater as all delays must be compensated by a matching pre-delay so this is not usually applicable to real time use. However it is a valuable tool in post-production and mastering.
MIDI	Musical Instrument Digital Interface, used both for performance note data and in effects units like the FX8000 for loading programs and adjusting parameters, usually under control of a sequencer.
Operating system	The program that a computer runs when switched on that allows you to run one or applications, and supplies services such as disk access to every application in a standard way.
Peak/RMS	Please see RMS/Peak
Power Linear	New method of using classic EQ in which both linear phase and extended control range are provided.
Program	Often called a "preset". This is a snapshot of the settings of an FX8000 unit saved on the internal hard disk for recall at a later date. Such a program may also be uploaded to a PC using the Replimat software for transfer to other machines or for archiving. A program also contains a list of effects or samples required to complete the effect, which are separately stored.
Process	Often used as an abbreviation for Replicator Process, or generally for the effect a processor has on a signal.
Replicator Process	The signal processing section of the system that executes the Dynamic Convolution process, which is responsible for replicating the effect of a sampled analogue system. This system requires a number of co-operating DSP chips to perform the necessary task.
RMS/Peak	Two measurements of signal level. Peak is the familiar maximum and minimum levels achieved by the audio signal waveform. RMS is a measure of the energy contained in the signal. Signals of the same peak level contain most energy if they are a square wave, half as much energy if they are a sine wave, and much less for spiky (and typical audio) waveforms. This is referred to as the "crest factor" and is the reason why a compressor that optimises peaks can sound very different from one which optimises energy.
Sample	A sample of how an audio processor affects the signal passing through it, taken in such a way that the unique Dynamic Convolution process can recreate the audio effect of the original unit entirely digitally, including any level dependent manifestations of the original, usually analogue, processor. Samples can be made on an FX8000 and are also used in the construction of Classic EQ and Classic Compressor effects.
Sample Rate	The sampling or clock rate of a digital audio system, typically 44.1kHz for CD, 48kHz for much master recording, and 88.2kHz or 96kHz for high sample rate recording. These sample rates are twice the maximum frequency that can be processed by a system. The higher sample rates are often thought to sound better but this can often be accounted for by simplistic digital systems (for example most digital EQ) suffering problems in the top octave. The higher sampling rates move this problem beyond the normal hearing range. Classic EQ for example does not suffer this effect as it accurately reproduces the original

<b>term</b>	<b>means</b>
	equipment right up to the maximum frequency possible.
Sampling	The process of passing a special test signal through an analogue processor, taking samples at multiple levels from peak audio level through a range of over 40dB to assess the performance of the processor and storing a set of derived responses as a sample for later recreation of the effect using the Dynamic Convolution process. This also includes in the case of a dynamics processor, measuring one or more of the gain reduction curves of the device for later simulation of the dynamic characteristic.
SPDIF	Sony Philips Digital Interface. Similar to AES but carried on an unbalanced cable or on a domestic optical fibre. An SPDIF wired signal can be connected to the FX8000 AES input using a suitable adapter and the FX8000 will receive SPDIF audio provided the signal source is located fairly close to the FX8000, or for sample rates up to 48k, the SPDIF may be fed to the coaxial or optical SPDIF input of the FX8000.

## Technical Specification

### Overview

- Channels 2 (standard), 4, 6 or 8 optional (may be fitted as upgrades).
- 2 – 8 channel compressor/equaliser running up to 96kHz sampling rate for high sample rate surround compression and EQ. (Channels may be used independently).
- Fully balanced professional input and output on both analogue and digital connections.
- 24-bit analogue conversion, 24-bit digital processing, up to 24-bit digital i/o.
- Single cable AES operation at all sampling rates using high speed digital audio transmitters and receivers.
- Word clock and AES clock reference in.
- Sample rates from 30kHz to 96kHz.
- High Impedance instrument input for direct processing and recording.
- Stereo mixed out available as well as clean feeds for each channel with full pan and level control (up to 48kHz).
- SPDIF interface for single stereo (2-channel) in/out (up to 48kHz) and compressor gain linkage between units (up to 96kHz).
- Optical interface supports SPDIF (stereo/2-channel) and ADAT multi-channel format (up to 48kHz).
- MIDI input and output for remote control from sequencer/keyboard and for remote operation.
- USB port for high speed interconnection to suitably equipped desktop computer.
- Internal hard disk for sample and program storage
- Balanced analogue input and output levels can be customised to meet individual installation standards by addition of plug-in modules.
- Processing latency: Typically better than 100 samples digital in to digital out (subject to change in the course of continuing development) plus any inherent delay in sample loaded as peak energy may not be at start of sample response.

### Channel Pair:

#### Inputs

##### 2 channel electronically balanced input on multiway connector:

+18dBu ( $\pm 1$ dB) for peak digital sine wave, >10k impedance.

##### AES balanced input (2-channel) on multiway connector:

Meets AES-3 1992 specification when wired to appropriate XLR connector using approved cable types. Also supports up to 96kHz 2-channel sampling as proposed for standard revision. AES embedded data is not processed or routed to output.

##### Unbalanced Analogue Input:

2–channels, professional phono (RCA jack) connectors.

+10dBu (2.46v RMS) ( $\pm 1$ dB) for peak digital sine wave, >10k impedance.

## **Outputs**

### **2 channel electronically balanced output on multiway connector:**

+18dBu ( $\pm 1$ dB) from peak digital sine wave.

### **AES balanced output (2-channel) on multiway connector:**

Meets AES-3 1992 specification when wired to appropriate XLR connector using approved cable types. Also supports up to 96kHz 2-channel sampling as proposed for standard revision.

### **Unbalanced Analogue Output:**

2 –channels, professional phono (RCA jack) connectors.

+10dBu (2.46v RMS) ( $\pm 1$ dB) from peak digital sine wave.

## **Master Section:**

The master section is limited in operation to 48kHz nominal sampling rate (or below). If you are using higher sample rates, you should use the main channel inputs and outputs (through an external mixer if needed). This is because some aspects, for example the ADAT® light-pipe interface is limited by design to 48kHz so this cannot be upgraded.

- The effects sampling outputs are not limited and are fully operational up to 96kHz nominal sampling rate.

## **Inputs (audio)**

### **SPDIF Stereo/2-channel Digital Input:**

Stereo (2-channel) Professional phono (RCA jack) connectors to IEC958. The signals on this input may be routed to any channel within the unit. This input may instead be used for compressor gain reduction linkage between units.

### **Optical input:**

This is switchable to receive either SPDIF data for consumer applications or to receive ADAT® multi-channel digital audio. Multi-channel formats are limited to a maximum of 48kHz sampling rate. The signals on this input may be routed to any channel within the unit.

### **Instrument Input:**

¼" instrument jack input, approx 50kHz impedance. Maximum input level for clipping 0.7V RMS. Minimum input level for peak digital audio (sine wave) –16dBu ( $\pm 2$ dB). Instrument sampling rate limited to 50kHz maximum.

## **Outputs (audio)**

### **Unbalanced Mixed Stereo Analogue Output:**

2 –channels, professional phono (RCA jack) connectors.

+10dBu (2.46v RMS) ( $\pm 1$ dB) from peak digital sine wave.

24-bit 64x oversampling converter .

### **SPDIF Mixed Stereo Digital Output:**

Stereo (2-channel) Professional phono (RCA jack) connectors and protocol to IEC958. This output may instead be used for compressor gain reduction linkage between units.

### **Optical output:**

This is switchable to carry either the SPDIF data for consumer applications or to carry ADAT® multi-channel digital audio.

### **Effects sampling output:**

**Balanced XLR** providing up to +18dBu from digital peak sine wave via 24-bit D-A converter.

**Balanced ¼" jack** – fully floating coupled output derived from the above signal but resistor attenuated 20dB. Connection can be wired with the ground lifted for direct connection to performance amplification or processing systems without generating hum loops.

**WARNING:** Only select high test levels in accordance with manufacturer's limits for device under test. Operating above maximum safe input levels may cause damage to device under test.

## **Data/Reference Connections**

**AES word clock input:** XLR to AES 1992 extended to 96kHz. System sample rate and AES frame may be synchronised to this signal.

**Word clock:** 75ohm BNC TTL level word clock input. System sample rate may be synchronised to this signal.

**MIDI in and out:** Allows MIDI program load and sequencing of major parameters.

**USB interface:** for connection to personal computers.

## **Digital Processes**

### **Total Processing:**

- 10 SHARC® floating point digital signal processor: one handling SPDIF and ADAT interfaces, monitor mixing and system administration., nine per channel pair. (Stereo system).
- 37 SHARC® floating point digital signal processor maximum (8-channel unit).
- Memory per 8-channel fully configured unit: greater than 208Mbytes.
- System Memory: greater than 16Mbytes.
- Memory per channel pair: greater than 48Mbytes.
- Maximum AFX Delays: Max 256 seconds (8-channel system, 48kHz) comprising 32 seconds (48kHz) 16 seconds (96kHz) per channel. Note AFX delays are available in multichannel groups up to 48kHz sampling and in mono and stereo groups up to 96k processing.
- Maximum Processing: 800Mflops per channel.
- Disk storage: not less than 3Gbytes per system.

## **MIDI: The gory details...**

The implementation is designed to be as transparent as possible. For more information about MIDI terminology there are many books available on the subject, or see the complete MIDI spec v1.0 available from the MIDI Manufacturers' Association.

### **Channel Selection**

Because an FX8000 has up to 8 channels, it is able to transmit and receive commands on more than one MIDI channel. MIDI channels map directly onto physical channels. For example MIDI channel 1 will control FX8000 channel 1. If the FX2000 is linked into stereo or multichannel groups, messages on any channel of a group will update the stereo pair together, and the FX8000 will transmit changes on the lowest channel of the group only.

The FX8000 will also transmit and receive program changes on MIDI channel 16 as a "whole machine load" which permits reconfiguration (e.g. stereo to mono and vice versa). (see "Program Load" below).

The FX8000 normally operates in OMNI OFF mode. This may be changed by a MIDI mode change message on the FX2000 basic channel which is 1.

### **OMNI OFF**

The FX8000 receives commands for channel 1 on MIDI channel 1 and for channel 2 on MIDI channel 2 (and so in if further channels are fitted). A command for a channel will apply to all the channels in that group.

Commands on other channels will be ignored. (except for channel 16 "whole machine load").

### **OMNI ON**

The FX8000 can be selected to OMNI ON by means of a MIDI OMNI ON message addressed to the basic channel (which is set to channel 1).

In this mode, a command on any channel will be applied to the channel currently selected on the front panel (or both channels if operating in stereo). A "whole machine load" addressed to channel 16 will accordingly be treated as a simple load of the current channel rather than what was intended (see "Program Load" below).

A MIDI reset command will switch OMNI OFF if MIDI is enabled.

## Program Load

The FX8000 accepts standard MIDI Program Load commands. These are usually preceded by a MIDI bank select command to specify which bank to load from. If this is missing, the last set bank will be used. This is not shown in the program load page as (for example) you may be working on channel 2 while a sequencer is controlling channel 1.

The normal mode is OMNI OFF and this will allow a new program to be loaded into either of the two channels if operating in MONO or into the channel pair if operating in stereo. If possible the program is loaded "seamlessly", i.e. if the model specified does not change and the routing is the same, a new set of parameters will be cross-faded to without an audible glitch. If operating in MONO the second channel will not be affected by this load.

If a program load is received addressed to channel 16 (and the mode is OMNI OFF), the FX8000 will treat the program load as a "whole machine load". This allows a program to reconfigure the FX8000 from between stereo, mono or multi-channel. This is always done with the FX8000 muted on all channels as it is not possible to perform a glitch-free change of the system configuration.

If the MODE is set OMNI ON a program load message on any channel will be directed to the currently selected channel(s). It is not possible to perform a "whole machine load" as the channel number for this (16) is ignored.

After a program load, the system will be busy for several seconds. Further MIDI commands issued during this period will be missed. If the command is from a sequencer for real time control, be sure to allow sufficient time including time for the FX8000 disk to spin up if it has timed out and stopped.

Bank select commands may be issued with either the LSB or the MSB MIDI bank select command. Since values other than 1 - 8 are ignored, a full MSB, LSB pair will work fine too. The FX8000 sends bank select as an LSB only.

## Parameter Control

Parameter control is provided to allow MIDI automation, or fine tuning, of an FX8000 program. It is not intended to give full control replacing the front panel functions. It can adjust the main EQ and dynamics controls in real time but cannot load models or configure the system. This should be done by means of loading programs.

The FX8000 can receive or transmit parameter control in two common formats. SYSEX messages can be used in which the 7 or 14-bit parameter value can be controlled easily by most sequencer software, or control change codes accessing "Non Registered Parameter Numbers" (NRPN's) may be used. Which of these you choose depends on the capabilities of your sequencer or MIDI control software.

The FX8000 can also transmit parameter changes when a control is altered (in either the SYSEX or the NRPN format). These are suitable for recording in a sequencer for automation on replay.

Most parameters are set using 14-bits of precision. It is possible to switch the FX8000 to receive only the MSB of such commands to allow control from MIDI controllers which only output 7-bit slider values, for example. Adjustment will of course be a little less precise.

For more details and a MIDI implementation chart see the end of this guide.

## MIDI echoes

When the FX8000 is connected to a sequencer it is not uncommon for MIDI messages to be echoed by the sequencer. This usually presents operation problems as due to the buffering in the sequencer and the FX8000 MIDI ports, a sequence of small changes can arrive delayed and interrupt the changes you are trying to make.



## Sintefex FX8000 Analogue Sampling Digital EQ / Compressor / Effects Unit

To prevent an echoed program load or parameter change from duplicating a front panel operation it is possible to set an "ignore" time, in which any MIDI messages received after a MIDI message is transmitted will be ignored for a short period.

This should be used if it is not possible to prevent the sequencer from echoing messages it receives. It does mean that messages already recorded - say for channel 1 in a two channel mono set-up - may be ignored as you record changes for channel 2, at least for the period set for ignoring messages.

The "ignore" period can be set simply to "on" to get a default ignore period (typically 1 second), or to a period from the list in the MIDI setup page. The period is automatically multiplied by ten for ignoring program load messages received, to make sure that a slow program load does not get repeated because the time has already elapsed when it is complete.

The FX8000 will not re-transmit a change it receives via MIDI, so "infinite loops" cannot occur.

## MIDI Implementation Chart and Parameter Numbers

Date: 24th March 2003,  
version:3.0.0

Model: FX8000

### MIDI Implementation Chart

Function...		Transmitted	Recognized	Remarks
Basic Channel	Default	1	X	Also recognises one MIDI channel per audio channel (OMNI OFF)
	Changed	1-16	X	
Mode	Default	Mode 3	X	Default is OMNI OFF. Poly and Mono are not relevant to this unit (note 7)
	Messages	X	OMNI ON/OFF	
	Altered	*****	POLY ignored	
Note Number	True Voice	*****	X	As this is an effects unit, note data is ignored. Hence also there is no meaning to Poly/Mono mode.
Velocity	Note On	X	X	
	Note Off	X	X	
After Touch	Key's Channel	X	X	
Pitch bend		X	X	
Control Change	0	X	O	Bank Select MSB (note 6)
	6	O	O	Data entry MSB (note 1)
	32	O	O	Bank Select LSB (note 6)
	38	O	O	Data entry LSB
	96	X	O	data increment (note 2)
	97	X	O	data decrement
	98	O	O	NRPN LSB (note 3)
	99	O	O	NRPN MSB
	100	O	O (127 only)	RPN LSB for "null" only
	101	O	O (127 only)	RPN MSB (note 4)
Program Change		O	O 0-127	in banks 1 - 8
	True Number	*****	O 1-128	
System Exclusive		O	O	(note 5)
System Common	Song Position	X	X	
	Song Select	X	X	
	Tune Request	X	X	
System Real Time	Clock	X	X	
	Commands	X	X	
Aux Messages	Local On/Off	X	X	
	All Notes Off	X	X	
	Active Sensing	X	X	
	System Reset	X	0	resets MIDI system only
Notes				reset selects OMNI OFF

Mode 1: Omni On, Poly  
Mode 3: Omni Off: Poly

Mode 2: Omni On, Mono  
Mode 4: Omni Off, Mono

O: Yes  
X: No

### MIDI Note 1: SYSEX versus NRPN Parameter Control

FX8000 parameters are specified by non registered parameter numbers according to the table (note 3). Remember that a MIDI byte is only 7 bits and a 14-bit value is sent as an MSByte and and LSByte (more significant byte and less significant byte).

Parameters may be updated by SYSEX messages or by use of NRPN controls. Which you use depends on which your sequencer supports best.

When using NRPN controls it is necessary to issue a parameter select message pair (an MSByte and an LSByte) followed by a new parameter value, usually also an MSByte and LSByte. This is usually followed by an RPN NULL message. It is important that the sequencer does not change the order of any of these messages, or mix them up with other control messages. On the other hand, SYSEX messages are single self contained messages and a sequencer cannot mix these up. The SYSEX format is shown below (note 5) and is fairly straightforward to edit or attach to controls in a many sequencers.

### **NRPN Format**

Most parameters need 14-bit DATA so require both an MSByte of data and an LSByte of data to complete the update. These will be updated whenever the LSByte is received, so the MSByte may be omitted if it does not change. When an MSByte is received the LSByte is set to zero.

Some parameters require only 7-bit DATA (switches, selections from lists etc) and these do not require an LSByte. These are updated whenever an MSByte is received.

In addition to this, it is possible to set the FX8000 so that all parameters are restricted to 7-bits (on reception only), in which case only an MSByte is required and any LSByte is ignored. This is for control by a MIDI controller that can output only 7-bit parameters, e.g. from a slider or knob.

Parameter numbers (NRPN's) also require an MSByte and an LSByte. Both these must be present to select a parameter but can occur in either order.

When a parameter is selected, several things happen:

- Any previous parameter will have its data updated by the previous parameter DATA value if that has not already happened (e.g. if an MSByte of data arrived but no LSByte - this is for transparent support of controllers that can only send 7-bit data).
- The value of the addressed parameter is loaded into a holding register for update. It is converted from its native format to a 14-bit value (or 7-bit for some parameters) in as natural way as possible so that 0 represents a minimum setting, 0x3fff represents a maximum setting, and 0x2000 (MIDI bytes 64,0) represent a typical mid position for the control.

The controlling system should then send MSByte, LSByte data pairs. After each LSByte is received the parameter is converted back to native format and the FX8000 is updated. (Parameters with only 7-bit range require just the MSByte).

The controlling system can also issue Data Increment and Data Decrement commands. These are applied and executed immediately. (See note 2).

When update of a parameter is complete, the controlling system can do one of several things:

- Select a different parameter with an MSByte, LSByte pair at which point the parameter control process is ready for a further update
- Cancel the parameter select - this is done by issuing an RPN of 127,127 (often termed a NULL selection). This is recommended. (See note 4)
- Leave the parameter selected. This is dangerous as it makes the FX8000 susceptible to unwanted accidental further update of the selected parameter.

A sequence of parameter select and modify messages must address the same channel. If a data entry is encountered for a different channel to the last parameter selected, the parameter will be marked as unselected and no further changes will have any effect until a new parameter control sequence is successfully received.

When transmitting parameter changes an FX8000 transmits and NRPN parameter selection bytes, then the parameter value byte(s), then issues an RPN NULL (127.127).

### **MIDI Note 2: Data Increment / Decrement**

Provided an NRPN parameter is selected, all parameters value will be incremented and decrement by one MSBit. This can be fairly coarse for 14-bit parameters but may help find an approximate setting quickly. Some parameters, e.g. stepped parameters such as on/off may require up to 64 steps to achieve a change.

### MIDI Note 3: Non Registered Parameter Number (NRPN) Assignment

Note that only parameters listed here are controllable with MIDI. For other controls you should save the settings as a program and use a program load command.

NRPN#	MIDI bytes	Name	MIDI range (hex)	Internal Range	Notes
1 - 127		reserved			
128	1,0	system bypass / enable	0 & 7f	off/on	note 1
129-255		reserved			
256	2,0	Compressor Enable	0 & 7f	off/on	note 1
257	2,1	Compressor Attack	0 - 3fff	0.1mS - 1000mS	mid is 64mS
258	2,2	Compressor Release	0 - 3fff	1mS - 10S	mid is 640mS
259	2,3	Compressor Slope	0 - 3fff	1:1 - limit	mid is 10:1
260	2,4	Compressor Threshold	0 - 3fff	-40dB - +20dB	
261	2,5	Compressor Ingain trim	0 - 3fff	-60dB - +20dB	
262	2,6	Compressor Outgain trim	0 - 3fff		
263	2,7	Compressor Curve	0 - 7f	hard - sampled	>=0x40: sampled
264	2,8	Compressor Rectifier Flags	0 & 7f	4 bits	
265	2,9	Compressor Link Gain	0 & 7f	off/on	note 1
266		reserved			
267-275		reserved			
276	2,20	Expander Enable	0 - 7f	off / expander / gate	<=0x40: off; 0x40 - 0x5f: expander; >=0x60: gate
277	2,21	Expander Attack	0 - 3fff	0.1mS - 1000mS	mid is 64mS
278	2,22	Expander Release	0 - 3fff	1mS - 10S	mid is 640mS
279	2,23	Expander Depth	0 - 3fff	-20dB - 0dB	mid is -10dB
280	2,24	Expander Threshold	0 - 3fff	-80dB - 0dB	mid is -30dB
281-511		reserved			
512	4,12	digital EQ enable			note 1
513	4,0	Band 1 level	0 - 3fff	min to max available	zero at 0x2000
514	4,1	Band 1 freq			mid at 0x2000
515	4,2	Band 1 Q	0 - 64 hex 0 - 9: shelf		0xa: Q of 1.0 0x64: Q of 10.0
516-518	4,3-5	same for band 2			
519-521	4,6-8	same for band 3			
522-524	4,9-11	same for band 4			
525-767		reserved			
768	6,0	Classic EQ enable	0 & 7f	off/on	note 1
769-800	6,1 - 32	Classic EQ controls, typically up to 32 available, though normally at most 12 are normally used.	0 - 3fff clipped when out of model range, mid at 0x2000 where possible	min to max available	controls are allocated according to each classic EQ model, (note 3)
801 on		reserved			

Note 1: For all on/off values, received values of 0x40 (decimal 64) and above represents on. The FX8000 transmits 0 for off and 0x7f (decimal 127) for on.

Note 2: Parameters numbers are split into MIDI MSBytes and LSBytes, i.e. 514 is (4,2). These may be received in either order. To avoid confusion a 127 in either byte is considered illegal and will set the FX8000 parameter number to invalid. This has the same effect as a RPN NULL (127,127).

Note 3: Because Classic EQ's can have various different configurations, there is no sensible mapping from one Classic EQ model to another. Ensure that you have loaded the correct classic EQ (by a preceding program load if necessary) for a MIDI sequence of control changes.

### Value Coding for Gain, Frequency etc

For programming remote control operations it is useful to know the way the actual value of parameters map onto the MIDI MSByte and LSByte values. These are shown here as for example (64,24) to mean a MSByte of 64 and an LSByte of 24. Each can range from 0 to 127.

The NRPN assignment table lists the centre value for most parameters. This is coded to the MIDI value (64,0) as this is the centre of the MIDI control range.

The MSByte is given a sensible range for parameters so that a single byte controller can be used for reasonable control of a parameter. The LSByte adds additional precision, often two extra decimal places by running in the range of 0 - 99. In cases like this any LSByte value above 99 is treated as 99. This allows MIDI controllers that offer two-byte control to select parameters to a much higher accuracy.

The table below shows the mapping used for the main parameter types. Refer to the NRPN assignment table for the centre value and the maximum and minimum values a parameter may have. Any message received outside the maximum or minimum will be safely limited to the range available.

parameter type	MSByte	LSByte controls	Notes
Gain (dB) (including level, threshold etc)	1dB steps above centre value (up to +63). 1dB steps below centre value down to -40, then 3dB steps to -112dB.	For 1dB steps this ranges from 0 - 99 for two decimal places of additional gain. For 3dB steps this range from 0 - 119, each step giving 0.025dB of additional gain	
Time (ms) with centre value up to 6.4mS	0.1mS steps from centre down to zero, and up to a value of 96. Above this it give 10mS steps.	Values 0 - 99 give two extra decimal places, i.e. 0.001mS steps below MSByte of 96, and 0.1mS steps above this.	
Time (ms) with centre value up to 64mS	1mS steps from centre down to zero, and up to a value of 96. Above this it give 100mS steps.	Values 0 - 99 give two extra decimal places, i.e. 0.01mS steps below MSByte of 96, and mS steps above this.	
Time (ms) with centre value up to 640mS	10mS steps from centre down to zero, and up to a value of 96. Above this it give 1000mS steps.	Values 0 - 99 give two extra decimal places, i.e. 0.1mS steps below MSByte of 96, and 10mS steps above this.	
Time (ms) with centre value above 640mS	100mS steps from centre down to zero, and up to a value of 96. Above this it give 10000mS steps.	Values 0 - 99 give two extra decimal places, i.e. 1mS steps below MSByte of 96, and 100mS steps above this.	
Frequency	64 gives 1kHz. Each MSbyte step gives ISO 1/6th octave frequency centres down to 1Hz or up to 100kHz	This provides 4 steps between each frequency step to give ISO 1/24th octave centres. The top 2 bits only select this, i.e. values 32, 64, 96.	
Slope	0, 10, 20,... map onto 1:1, 1.5:1, 2:1, 3:1, 4:1, 5:1, 8:1, 10:1, 12:1, 20:1, 25:1, limit	Only used for 0.1 steps in the larger ranges	note 1
Curve	up to 6, hard knee, up to 18, soft knee 1 etc. in intervals of 12. From 54 and above: selects sampled curves	not used	
Q	10 - 100 selects 1.0 to 10.0, below 10 selects shelf	not used	

note 1: LIMIT is assumed for any value above 25:1 and is selected by an MSByte of 110. Values above 110 are currently not used but may be assigned in the future. Intermediate values select steps in size of 0.1. This does not require the LSByte up to 5:1, but above this the steps of 0.1 include the LSByte. These higher precision steps are only seen where the slope control is selected to be fully variable rather than the more normal switched setting

#### MIDI Note 4: RPN Null

An RPN "Null" (127,127) will clear the currently selected parameter so that no parameter is selected. If a parameter was selected it will be updated by the current value of the parameter so that if only an MSByte was received for a 14-bit value a "course" setting will be performed.

#### MIDI Note 5: SYSEX Support

There are two formats of SYSEX command.

One set has been implemented for factory testing and for initial software load only using proprietary software applications (e.g. Replimat PC software). These are not suitable for general use.

The other set is for parameter control as an alternative to NRPN control change messages. The message format is as follows:

<b>byte (decimal)</b>	<b>meaning</b>
0xF0 (240)	sysex start
0x00 (0)	vendor id #1
0x20 (32)	vendor id #2
0x3D (61)	vendor id #3
0x6n or 0x70 (96 - 112)	n is machine select 0x0 - 0xf OR all machine select 0x10. Machine ID is the same as the basic channel. (note 1)
0x00 - 0x07 (0 - 7)	channel select
0x00 - 0x7E (0-126)	parameter select MSB (note 2)
0x00 - 0x7E (0-126)	parameter select LSB
0x00 - 0x7F (0-127)	parameter value MSB
0x00 - 0x7F (0-127)	parameter value LSB (note 3)
0xF7 (247)	EOX end of sysex

note 1: Other 0x7x values than 0x70 are invalid and will be ignored. They may be used for future further messages. Machines are currently set to number 0x0 (basic channel 1).

note 2: MSB is the upper 7 bits and LSB the lower 7 bits of a 14-bit value.

note 3: If no LSB is present (i.e. EOX occurs here, a value of 0 is assumed for LSB).

The parameter number and range is the same as for NRPN parameter setting.

## MIDI Note 6: BANK SELECT

FX8000 accepts bank select commands for either the LSB or MSB of the bank. Either are treated as setting the internal bank provided they contain the value 1 - 8. All other values including zero are ignored. The FX8000 transmits an LSB bank select when the user selects a program, prior to sending a bank load command.

## MIDI Note 7: OMNI OFF/ON

The OMNI mode does not affect the FX8000 MIDI transmission. It affects the MIDI reception as follows

<b>message received</b>	<b>OMNI OFF (default)</b>	<b>OMNI ON</b>
parameter set	this affects the audio channel indicated by the MIDI channel (actually affects the whole group of channels for stereo or multi-channel groups)	a parameter set with any MIDI channel is applied to the FX8000 channel(s) currently selected on the front panel
program change, channels 1 or 2	this affects the audio channel indicated by the MIDI channel (actually affects the whole group of channels for stereo or multi-channel groups)	a program change on any MIDI channel is applied to the FX8000 channel(s) currently selected on the front panel
program change, channel 16	performs a whole machine load of the selected program, including all channels of the FX8000 and any configuration changes in the program, i.e. changes of channel grouping for mono, stereo or surround.	the program change is applied to the FX8000 channel(s) currently selected on the front panel. A "whole machine load" is not performed and a configuration change is not possible.
bank select	the bank is set for any subsequent program load	the bank is set for any subsequent program load
mode change: OMNI on	changes to OMNI ON provided it addresses basic channel	no change
mode change: OMNI off	no change	changes to OMNI OFF provided it addresses basic channel
system reset	all MIDI buffers are cleared	reset to OMNI OFF and all MIDI buffers cleared
SYSEX messages	OMNI mode is ignored, a SYSEX parameter change will affect the channel it is addressed to in its data stream.	
other messages	ignored	ignored

### **MIDI Note 8: Running Status**

This software version does not use running status when sending messages. This improves the chance of a connection working if the cabling is interrupted and a status byte is missed, because it could be a long time before a new status byte is transmitted in a control change sequence.

However the FX8000 will receive messages with running status according to the MIDI v1.0 specification.

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