DPS24

DIGITAL PERSONAL STUDIO

OS Version 2.00

Quick Technical Reference
for operation in Live Applications

Version 1.0 (created and maintained by Simon Jenni)

Block Function Diagram for Input, Output and Mixer Section in Surround Mode of OS 2.00

April, 2006

Introduction

What it is: This Quick Technical Reference serves as a supplement to the Quick Technical Reference for OS Version 1.6 (1.60 and 1.61). It only shows the function in a surround project in the OS Version 2.0. The OS V. 2.0 also handles stereo projects, which it treats the same way like OS 1.6 does. The pages "Operating the controls" in this document only shows the additional controls (or their differences) in a surround project. You have to look up the other controls in the Quick Technical Reference for OS 1.6.

Bugs: OS Version 2.00 is not as stable as OS Version 1.60 is. So you have to take it as a Beta-Version (Be aware of: OS Version 1.61 is even worse, whereas OS Version 1.5074 works best for some DPS24 users).

My advice: Use ***OS Version 1.60 for your stereo projects*** and only switch to ***Version 2.00 to mix a surround project***. After you are done with the surround project, *switch back to OS Version 1.60* (here: switching means uploading the Firmware).

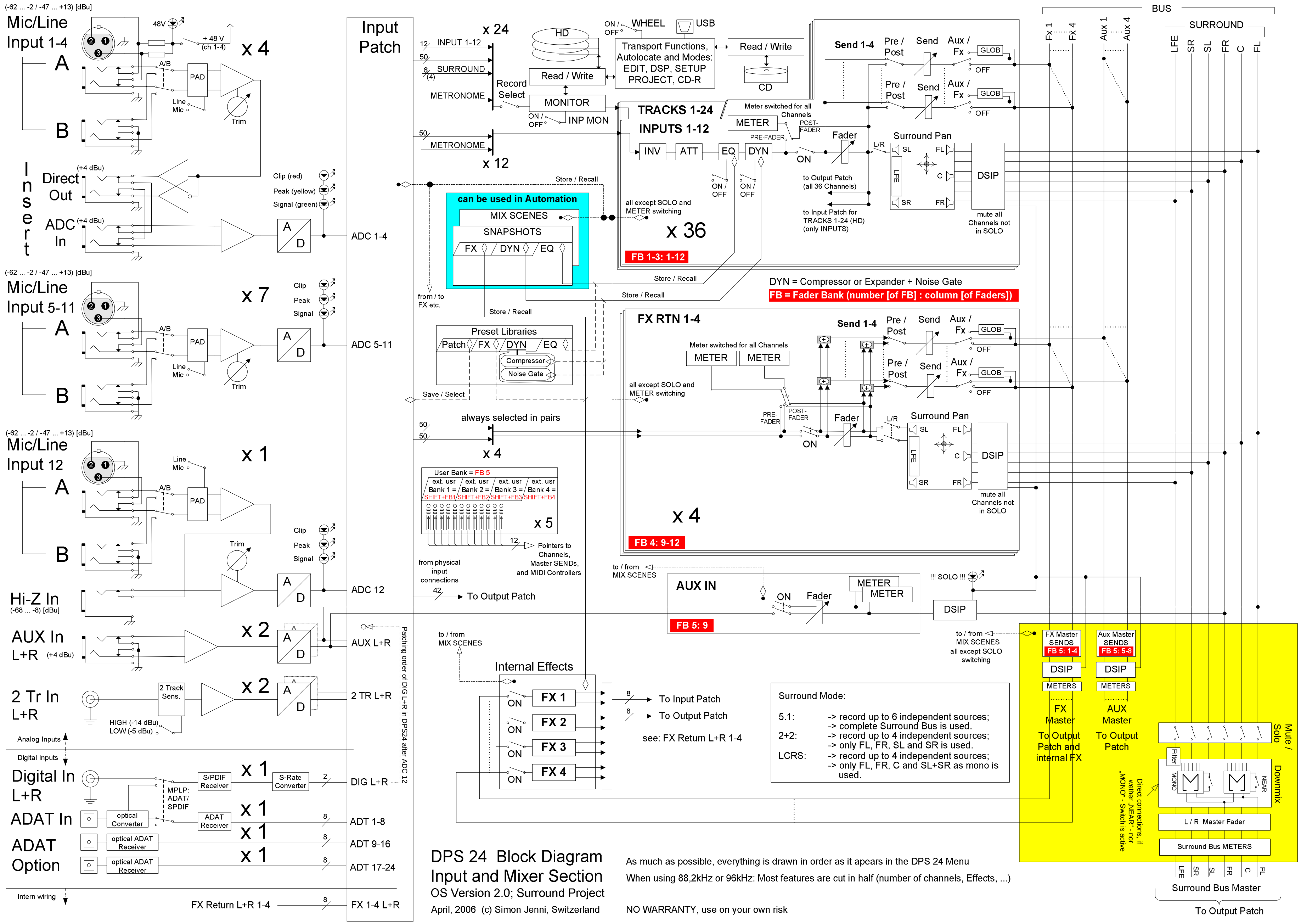
Important bugs I stumbled on when using OS Version 2.00:

- In several project-loading-procedures the DPS did hang showing forever "Loading *projectname*" (happened to me in both cases: loading a stereo or surround project). If this happens, you have to switch the DPS off and on again.
- It can happen, that after turning the DPS on, it shows forever "Scanning for drives ...". Also in this case, it did help switching the DPS off and on again.
- It even can corrupt your project. Switching off and on again will load the last good files.
- STEREO Project: If you solo a FX RTN Channel as PFL, AFL, or SIP, you will only hear the left channel. (I did not really work with stereo projects in V. 2.00, so there may be a lot more bugs).
- SURROUND Project: If you change an Aux and/or FX routing in a channel, you will see the meters of the Aux or Fx SEND in the wrong position. For instance: When I was switching (GLOB) FX 1 of a channel to Aux 1, the meter was shown on Fx 3 SEND. You can see this on the mixer page or channel of the FX Master SENDS in Fader Bank 5. You will see the meter in FX Master SEND 3 (Fader Bank 5: 3), but you will need to use the Fader of Fader Bank 5: 5 to influence the amount of send (and you will see no meter in there). Same happened when switching from Fx 2 to Aux 2 (you will see the meter shown on Fx 4 SEND) or from Fx 3 to Aux 3 and so on. However, after changing all channels and then reloading the project, the meters are shown in the correct place.
- SURROUND Project: Even so the Group bus is no longer used, they still show up in USER BANK ASSIGNMENT - Setup.
- SURROUND Project; Output Patching: You still can patch the SOLO – Bus. However, this bus is not used in surround mode (Only DSIP can be used). Therefore, SOLO L and R should not be patchable.
- SURROUND Project: You still are able to switch the Headphone Level HIGH, MID or LOW. However, there is no effect in the Headphone out in surround mode (in a stereo project, this will have the expected effect).
- SURROUND Project: Monitor Level does affect also the Headphone out. If you use a stereo project, the Monitor Level does <u>not</u> affect the Headphone out (This on the same DPS24).

➔ I did use Version 2.00 only for checking the drawings and the functions shown in this Quick Technical Reference. Therefore, there may be more bugs and flaws.

Have a lot of fun with your DPS24!

Greetings, Simon



DPS 24 Block Diagram
Input and Mixer Section
OS Version 2.0; Surround Project

April, 2006 (c) Simon Jenni, Switzerland

As much as possible, everything is drawn in order as it apears in the DPS 24 Menu

When using 88,2kHz or 96kHz: Most features are cut in half (number of channels, Effects, ...)

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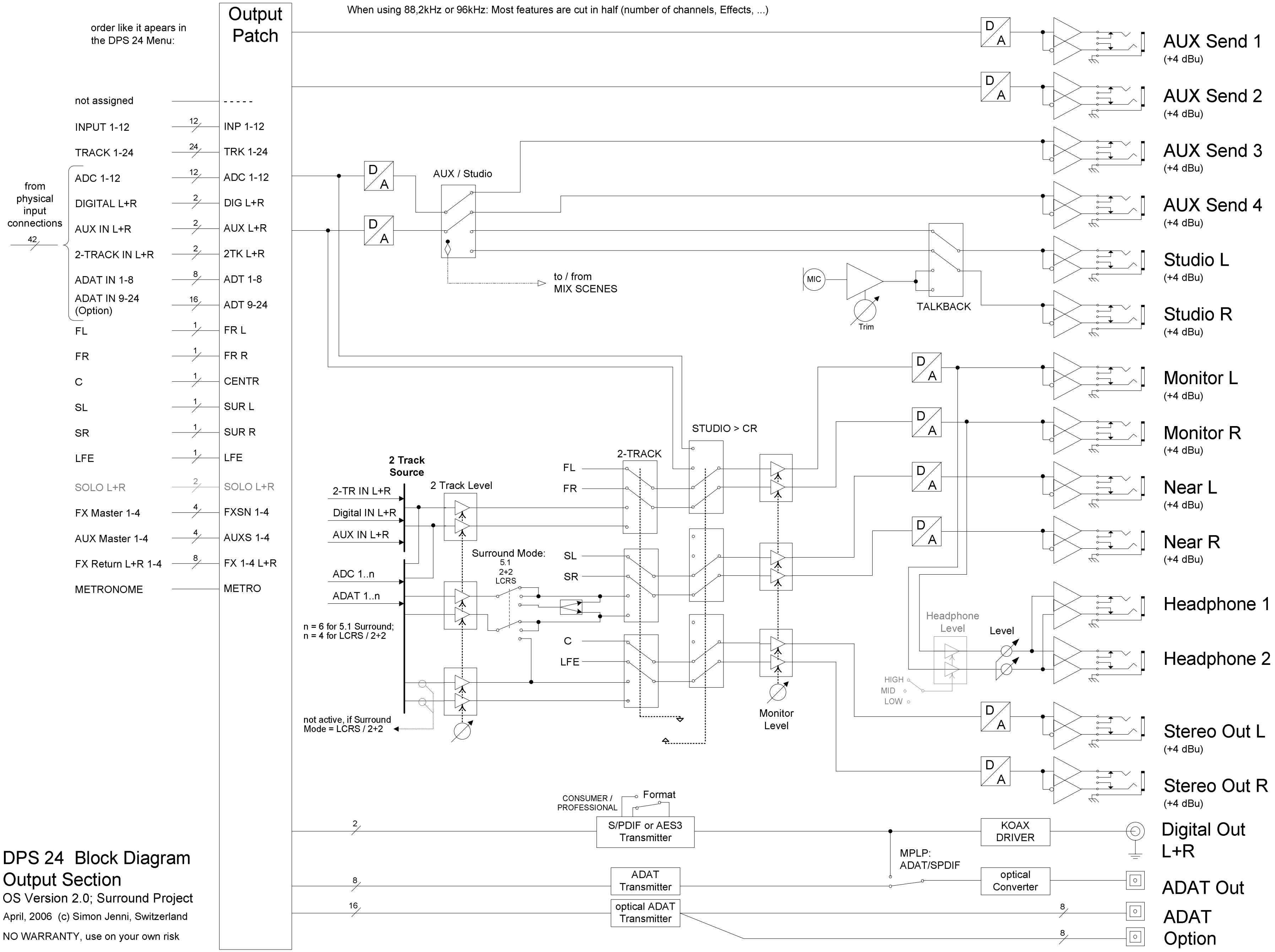

DPS 24 Block Diagram
Output Section
OS Version 2.0; Surround Project
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DPS 24 Block Diagram: Surround Pan of Mono-Channel's

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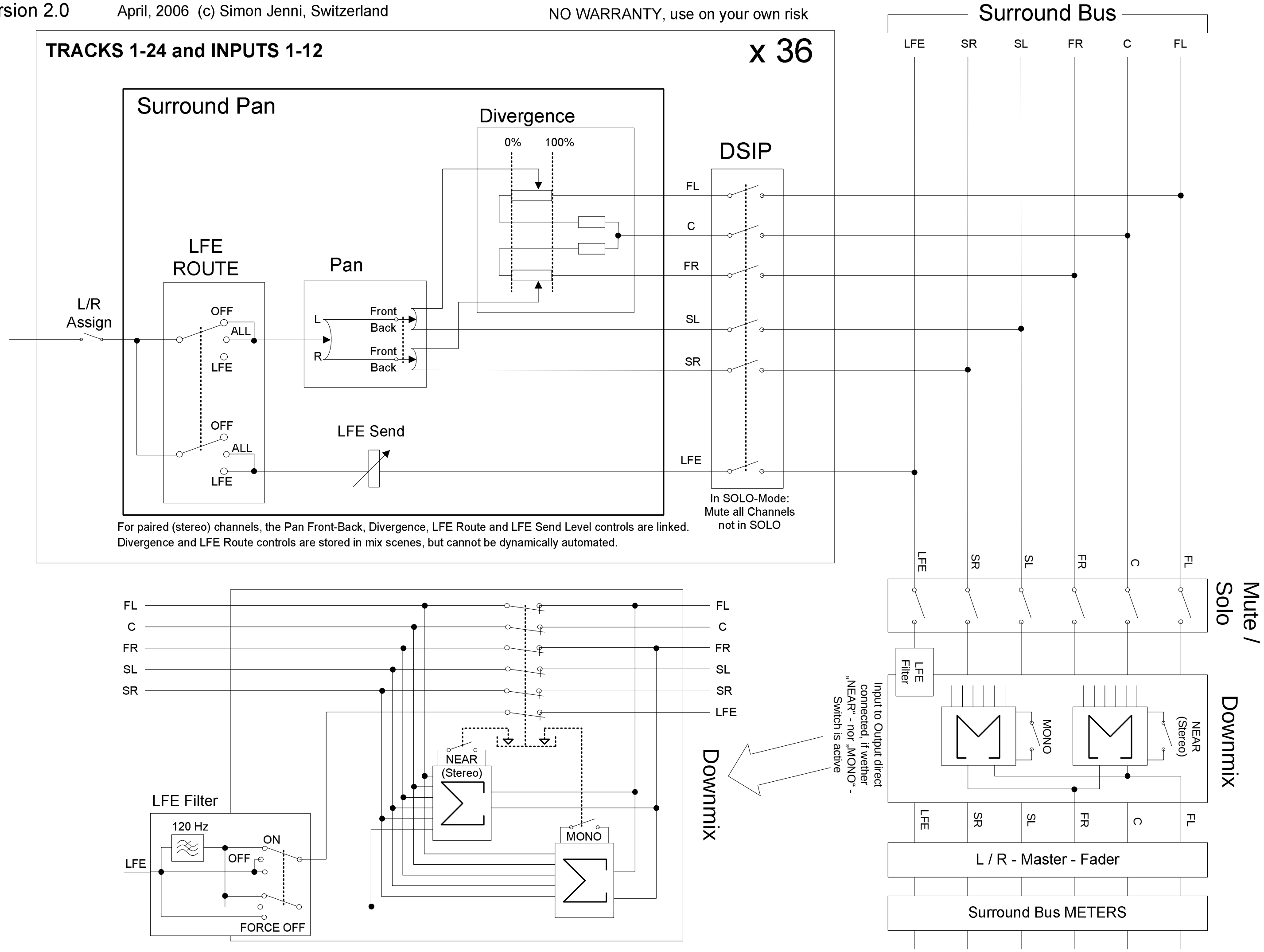

DPS 24 Block Diagram: Surround Pan of FX Return Channel's

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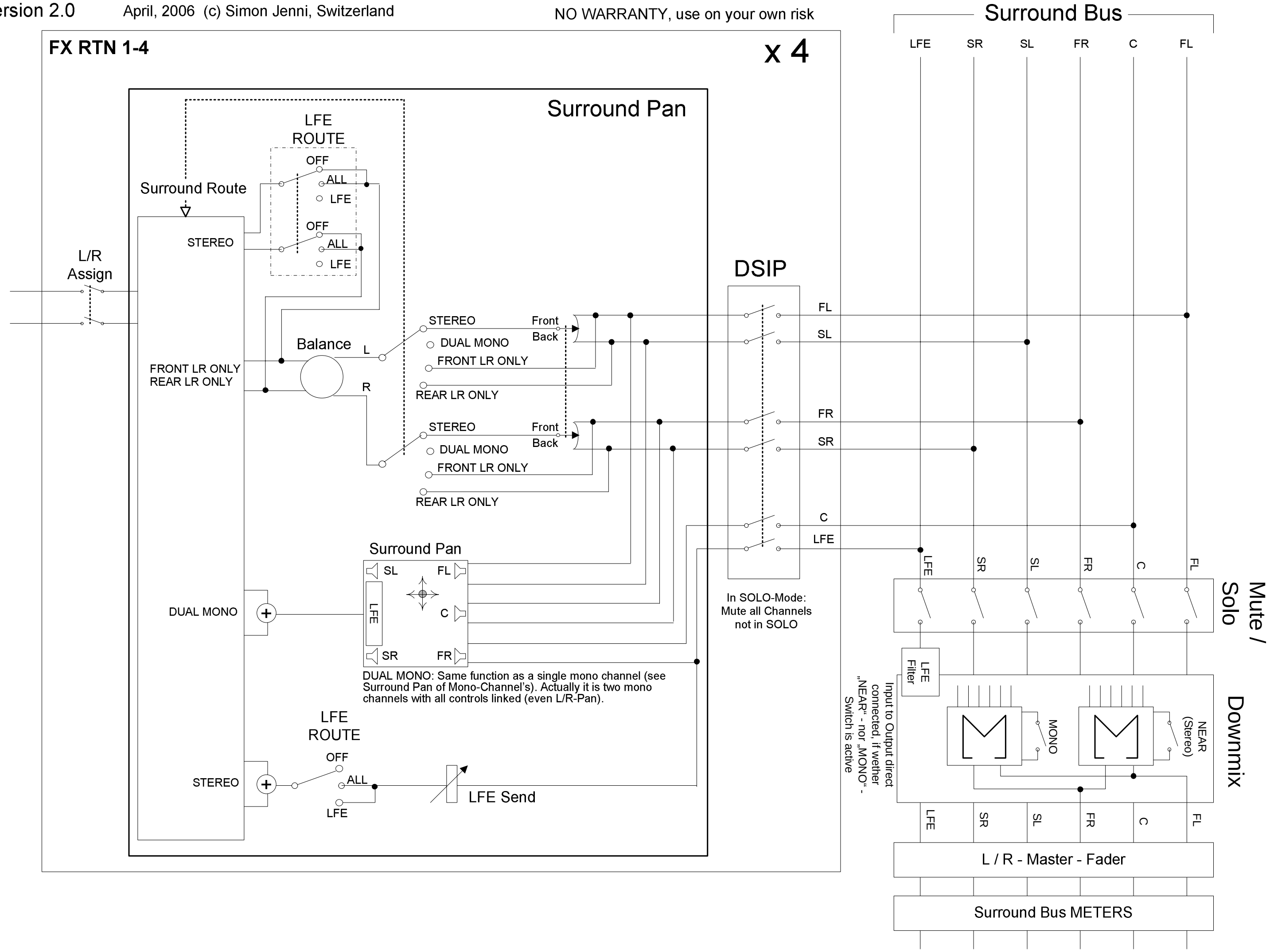


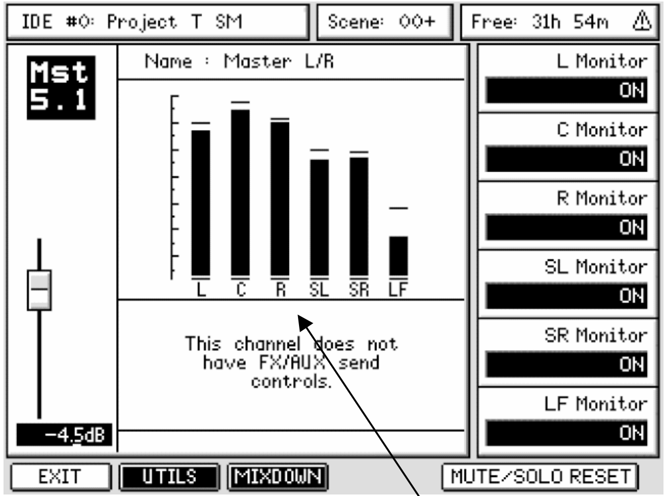
Operating the controls (only differences to OS Version 1.6)

Control-area abbreviations: see in Quick Technical Reference of OS Version 1.6

Control	Operation
SURROUND PROJECT Surround Mode LFE Filter	Any new project that you create with this version will be a Stereo Project by default. In order to mix your project in surround, you will need to convert it to a surround project. This operation is performed in the new MIXER MODE page. In MIXER mode, a new MODE.SET softkey (F4) is available: To create a SURROUND PROJECT: MIXER (MK=Mode Key) -> MODE.SET [F4] (LCD SK=Soft Key) -> CONVERT TO 5.1 SURROUND [F2, F3, or F4] (LCD SK) -> name the project -> OK [F6] (LCD SK) -> RESET MIXER [F5/F6] (LCD SK) {or may be you will use COPY CURRENT [F3/F4] (LCD SK)} -> Pan Mode [Q1] (LCD QL=Quick Link knob) to select a surround mode (5.1, LCRS or 2+2). If 5.1 is selected as Pan Mode, you can set up the LFE-Filter with [Q2]: -> LFE Filter [Q2] (LCD QL) to select a Filter Mode (ON, OFF, FORCE OFF).
24 Track Record	You can not use 24 Track Record in surround mode (only 6 in 5.1 and only 4 in LCRS or 2+2 mode).
Routing	Instead of STEREO L+R, GROUPS and SOLO you can patch the SURROUND bus components in TRACK-Channels and Output Patching. Output Patching of MAIN L+R, NEAR L+R and STEREO L+R are fixed.
SOLO	Only DSIP (Destructive Solo In Place) is available.
NEAR [Indicator: flashing when on]	3rd key from top in MM=Master and Monitor section Fold-down of surround-mix to stereo. Output only to Main L+R.
MONO [Indicator: lit when on]	2nd key from top in MM Fold-down of surround-mix to mono. Output only to Main L+R.
Mute / Solo	Go to MASTER CHANNEL page: -> Make sure that Q-Channel[1] is active = illuminated (ULC=Upper Left Control area, 6th key in last row). -> Press the SELECT key of the L/R (Surround) Master Channel (5th row of keys in MM). -> Use Q1 to Q6 (LCD QL) to solo or mute the surround channels. -> Use F5 or F6 (LCD SK) to global Mute/Solo Reset.

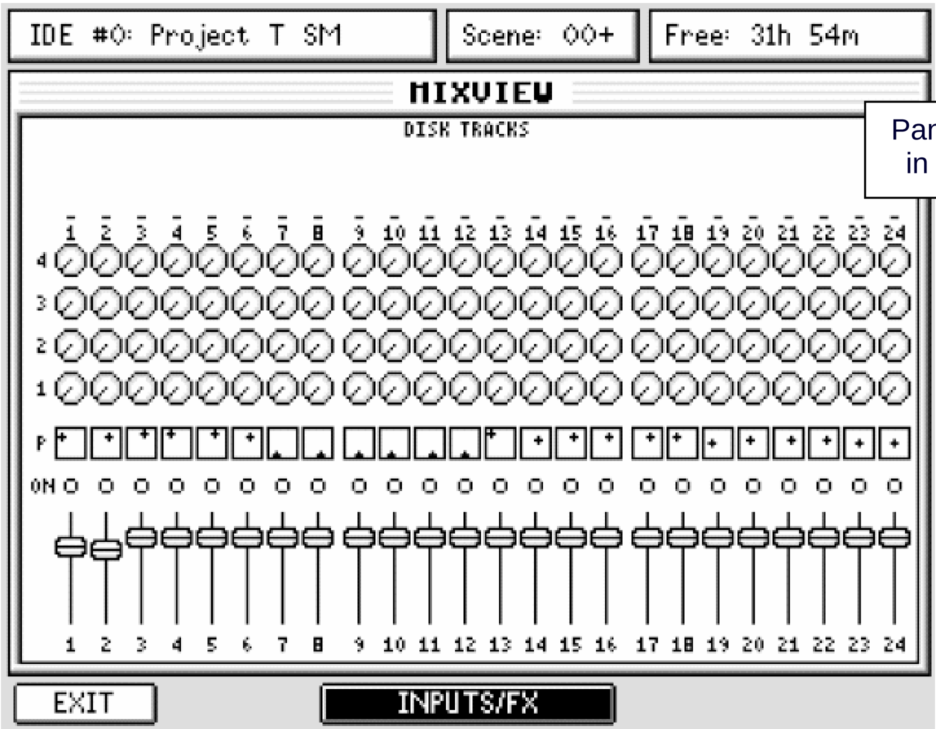
[1] If Q-Channel is active and you don't see the channel information in the LCD-Display: Hit the key twice.

SURROUND PANNING Pan, Divergence, LFE Send LFE ROUTE Using Pan Front-Back Flip [Indicator: PAN key is flashing when Pan-Fader flip is on] Surround Route	Go to the MIXER CHANNEL PAN subpage: -> Make sure that Q-Channel[1] is active = illuminated (ULC 6th key in last row). -> Use Fader Bank (FB in LRC=Lower Right Control area) to get to the channel-bank you need and then press the SELECT key (first row of keys in LLC=Lower Left Control area) of the channel you want. -> Use Q1 to Q4 (LCD QL) to set the appropriate parameter (Pan Left-Right; Pan Front-Back; Divergence; LFE Send Level). -> Use F5 or F6 (LCD SK) to set the LFE ROUTE (OFF, ALL or LFE). -> Make sure that Q-Channel[1] is <u>not</u> active = key is <u>not</u> illuminated (ULC 6th key in last row). -> Press the Q-Strip Function PAN key while PAN is the selected Q-Strip function. -> Use the faders (Full Back = fader down, Full Front = fader up). If you have selected a FX Return Channel there is one more control: -> Use Q6 (LCD QL) to set the Surround Route to STEREO, DUAL MONO, FRONT LR ONLY, or REAR LR ONLY.
2 Track Source [Indicator: lit when on]	You can also route ADC 1..n and ADAT 1..n as 2 Track sources (see function diagram).
Monitor Level	In surround mode this level affects the Monitor L+R, NEAR L+R, Stereo Out L+R and the Headphone out level.
Software Headphone Level	You still can switch this level to HIGH, MID, or LOW however it will not affect the Headphone out level.
Indicator	**Description**
PAN (SURROUND Position)	Mixer View: -> MIXER (MK) -> MIXVIEW [F2] (LCD SK): You will see a small surround pan placement graphic for each channel. -> use the softkeys to go to the appropriate channels. Channel View: -> Press Fader Bank (FB in LRC) of channel to view -> press SELECT key (first row of keys in LLC) for the channel to view -> activate Q-Channel (so it is illuminated) (ULC 6th key in last row): You will see the Pan placement graphic in the channel pan page.
METER (SURROUND)	MIXER (MK), MAIN (MK) or In MASTER CHANNEL: -> Make sure that Q-Channel[1] is active = illuminated (ULC 6th key in last row). -> Press the SELECT key of the L/R (Surround) Master Channel (5th row of keys in MM).



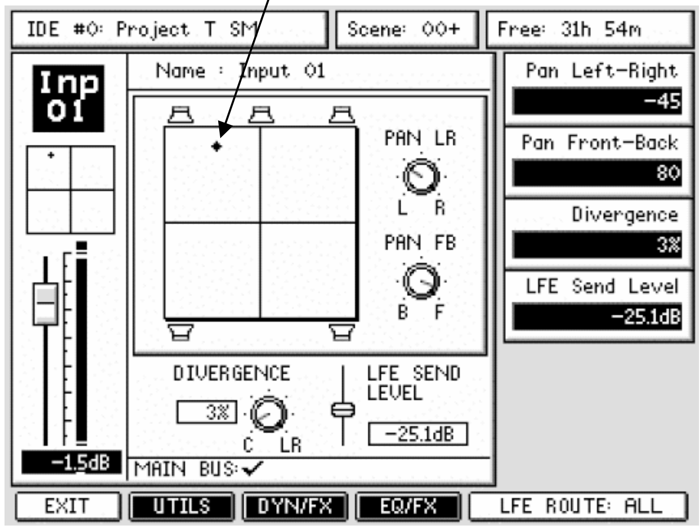
Surround Meter in Mixview

Surround Meter in Channel View



Pan Position in Mixview

Pan Position in Channel View



To Do's

Besides the elimination of the Bugs, there may be some reasonable enhancements for the next OS.

Features that can help a lot and don't should take to much an effort to implement:

- Make the four sends of each channel routable to all FX- and AUX- busses (every send should be able to switch to FX1, FX2, FX3, FX4, AUX1, AUX2, AUX3, or AUX4 and not only e.g. to FX1 or AUX1 and so on). It should be possible to use one send for FX1, another to AUX1 and so on. Leave it to the user not to assign more than one send to the same bus.
- Make the FX Master and AUX Master also available for routing in Input Patch for all Channels.
- Make the TRACKS 1-24 also available for routing in Input Patch for INPUTS 1-12.
- Make the Groups in stereo projects also available for routing in Input Patch for all Channels.
- Make the AUX IN Channel available for Input Patch routing or at least to switch between the physical inputs AUX IN L+R, 2 Track IN L+R, or Digital IN L+R.
- Make it possible to switch the DYN in each channel individually pre or post EQ.
- Make the Channel EQ 's as three full parametric EQ 's. Use therefore the already existing Q-fields in the Channel EQ-subpage. You can then navigate with Q5 (LCD QL) to the appropriate field and use Q6 (LCD QL) to change the Q-value. For the row "LF" in column "Q" you could show **LSHF** for low shelfing function and the Q-value for the Q-EQ-function. For the row "HF" in column "Q" you could show **HSHF** for high shelfing function and the Q-value for the Q-EQ-function. Leave the row "MF" unchanged (only Q-function can be used for this filter). The switching between shelfing and Q-filter should take place when the Q-value is turned to its lowest value (or lower than its lowest value).
  Since both algorithm (Q-filter and shelfing filter) are implemented, this has only to do with arranging them in here.

Features that will take some effort to implement, but would be really useable:

- Make the Channel EQ 's even better. Add the possibility to switch the Low and High Filter also to a Low Cut (high pass) and High Cut (low pass) filter. The switching to Cut-function should take place when turning the "Q-value" even further down (turning the Q-value of the row "LF" or "HF" to its lowest value and then turning down some more with Q6 switches from Q-function to shelfing function and then turning down even more switches from shelfing function to cut function). Show **LCUT** or **HCUT** in the Q-field of row "LF" or "HF", when the filter is in cut function. The GAIN-Field of the rows "LF" and "HF" would show the slope in the case of cut function. Either this may be a fixed value or even changeable. The FREQ-Field should still be changeable. Since the algorithm of a low-cut filter (same as a high-pass filter) and high-cut filter (same as a low-pass filter) are not yet programmed in the actual OS, this may take some effort to implement.
- Add a SHIFT-Q-CHANNEL – function, which would change the function of the PAN rotary key (in Q-CHANNEL-function) to set the Q-value for the low full parametric filter and the FX/AUX1 rotary key (in Q-CHANNEL-function) to set the Q-value for the high full parametric filter (both rotary keys with the switching between Q-, shelfing-, and cut-filter-function).

Features that will take a lot of effort to implement and probably pushes it somewhat to far (However, these features should still be possible to implement with the actual hardware-design of the unit; since there is no extensive (or fancy) matters asked for like e.g. VST-plug-ins, which really would require a new unit-hardware-design):

- Add an additional switchable low-cut filter to every channel.
- Add a hard-knee / soft-knee switch to the compressor.
- Add a switch, which makes the compressor sensible to RMS or peak.
- Make it possible, to use an audio signal of any AUX-bus, TRACK-, or INPUT-Channel as the input to the side-chain of the dynamic processor (compressor, expander, gate) on any other TRACK- or INPUT-Channel.
- Make it possible to switch up to two of the internal FX's (switchable "in parallel" / "in serial") as insert to the Stereo L/R bus (switchable "Pre"- / "Post"- Fader).
- Opposite-EQ-Channel-Link: The EQ of the linked channel sets up the parameter as opposite to the other channel, so that the sum would be like a flat EQ. This could be used for setting up a frequencies-sensitive compressor for e.g. a two band multi-compressor or a simplified DE-ESSER.
- Make it possible to use up to 8 SENDS in a channel using up the ones not needed in other channels (actually using up the processing power equally for this task). There should be a message telling when all SENDS are used up and don't let the user select more.
- Make it possible to use many full-parametric filters in a channel using up the ones not needed in other channels (give a message, when all channels are used up and don't let the user select more than are possible).
- Include a software dolby digital encoder to be able to finish a 5.1 surround sound CD or even DVD (then you would need to add also a DVD-driver).

If Akai is not interested in developing any more in the DPS24, there should be no problem for them to make the software development kit for the DPS24 OS downloadable for everyone (including the source code for at least OS Version 1.60 and 2.00). Then, it would be possible to add some features by other software developer using the DPS24. It would be clear, that using such self-developed OS would be the risk alone of the one using it.

Since this SDK would be so hardware specific, Akai would not have to fear that any competitors could draw some benefits from that.