3. Internal Structure of the RM3200D

3.1 DSP Processing and Routing

The DSP Processing inside the RM3200D consists of the following main DSP software parts:

- Input and Output Routing
- Input and Output Gain (includes also headroom and deemphase processing)
- Summing Busses (includes PFL and Conference/Mix-Minus)
- DSP Processes (Input Processing and Fixed Processing)
- Output Functions
- Monitor Busses

You will find detailed descriptions of the software parts in this chapter.

The Routing, the Input/Output Gain Processing and the Summing Busses are completely independent from each other and they are always processed.

Only the DSP Processes and the Output Functions share the same DSP power. The DSP power is displayed in the configuration software during the configuration process in the System Configuration Window. You will get an error message in the configuration software when you try to configure more than 100% of the available DSP power. In that case it is not possible to generate a downloadable configuration.

The Monitor Busses are special routing functions (no special DSP resources!) to build monitor selectors, and you can route out the Monitor Busses to a output.

The internal format of the audio data inside the RM3200D is 32 Bit Floating Point.

3.1.1 TDM Bus and Routing

In this section we describe the routing possibilities of the system. All input signals are converted to the floating point format after analog to digital conversion or after decoding the AES3/EBU or MADI input signal and the opposite processing on the outputs.

After floating point conversion, the headroom and the input gains are always processed. When using the Digital Input/Output Module RM330-111, the deemphase will be processed when activated on the source (CD or DAT). After this, the signal goes to the TDM Bus of the system.

From the TDM Bus you can pick up the signal directly to an Output Module or you can route the signal to an input of a DSP Process, a Summing Bus (Fader Channel Input) or an Output Function.

The routing operates completely without interference, i.e. there is no limited “routing power” in the RM3200D.

The largest RM3200D DSP Frame RM330-053 has a TDM Bus with 432 audio signals or “Time Slots”:

- 24 input slots, each with 4 input channels -> 96 Time Slots
- 3 MADI slots, each with 56 input channels -> 168 Time Slots
- 42 summing busses -> 42 Time Slots
- 54 output functions -> 54 Time Slots
- 36 Stereo DSP Processes -> 72 Time Slots

The time slot number or the address of the TDM Bus for the 432 signals is always the same.

The following figure 29 shows a typical signal flow through the system. Please study this example carefully, it is of key importance to understand the configuration process with the configuration software `DEFINITION.EXE`:

- The signal flows from an input module through the fix point to floating point conversion, through the headroom and digital input gain stage and through the optional processed deemphase to the TDM Bus.
- All inputs are processed that way. The deemphase will be processed only when using the Digital Input/Output Module RM330-111 and if the bit is set in the professional or consumer data stream. The deemphase processing is not available with the Digital Input Module RM330-110 (because there is no control connection between the module and the DSP) or any other module (RM330-120/121/421).
- It is possible to route the input directly to an output (input A to output D) or to the Input DSP Processing (input A). After the Input Processing, the signal flows to the inputs of the summing busses (to fader 1).
- An input signal without any Input Processing flows directly from the input to a summing bus (input B to fader 20).
- The output of the Summing Busses is also available on the TDM Bus and can be routed out directly (to output E) or to an input of an Output Function.
- Also an input signal can be routed directly to an Output Function (input C).
- The output of the Output Function is available on the TDM Bus and can be routed out directly (to output F) or to the input of a Fixed Processing function like the Limiter.
- The signal flows at least from the output of the limiter to the TDM Bus and from the TDM Bus to an output module (to output G).
- The signal flows from the TDM Bus trough the Output Headroom stage and through the floating point to fix point conversion stage to the output module.

Important Note: All other combinations of the routing are possible in the system!

You can use the system also only as a Routing Switch with the advantage to control the headroom, process DSP functions like Limiters or Level Detectors and to process a format conversion between analog, digital and multichannel signals.

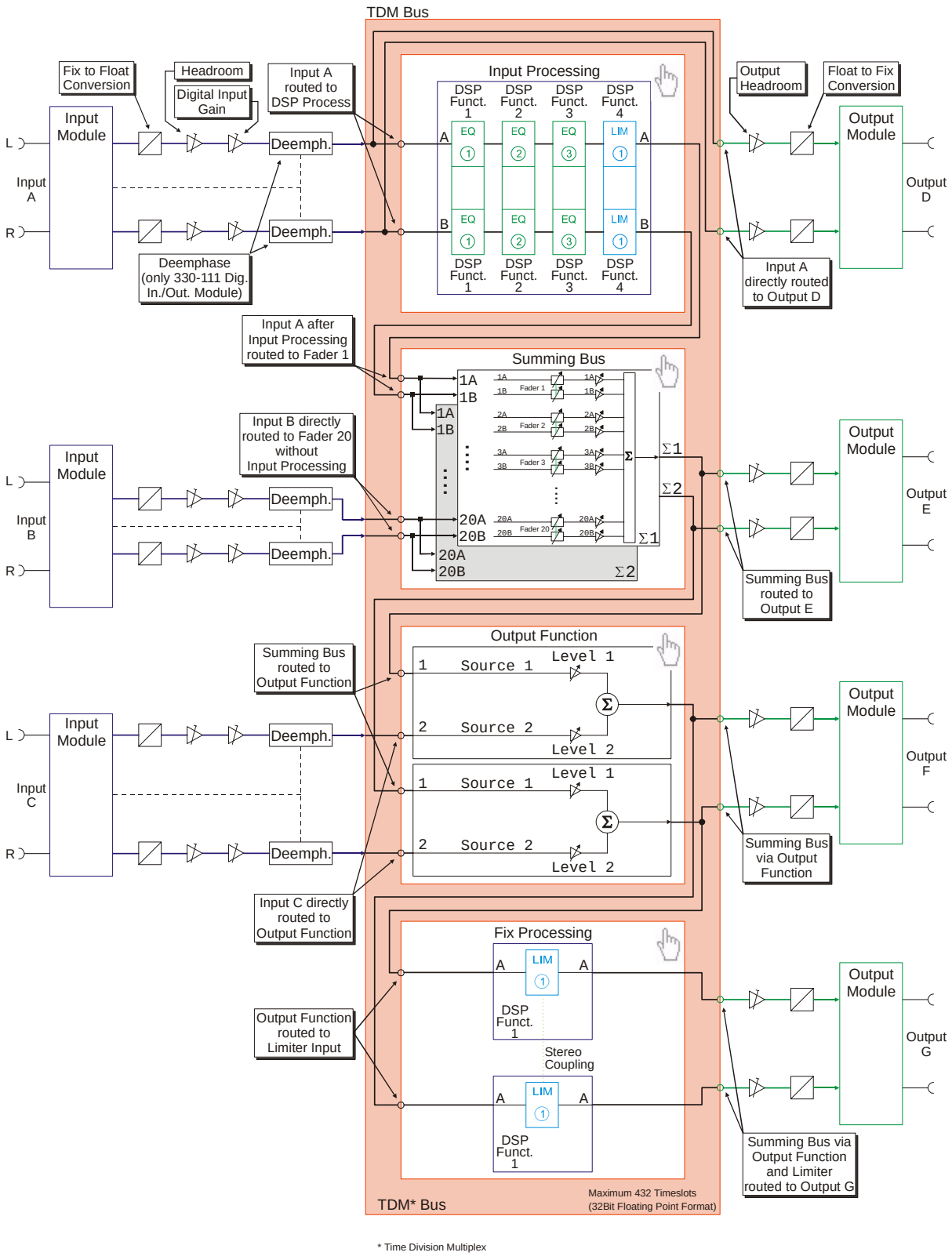


Figure 33: TDM Bus Routing

3.2 DSP Frame Types

Important Note: Like the different numbers of possible modules in a DSP Frame also the DSP power of the system is dependent on the size of the DSP Frame. The 30 slot frame has more processing power than the 20 or the 10 slot frame.

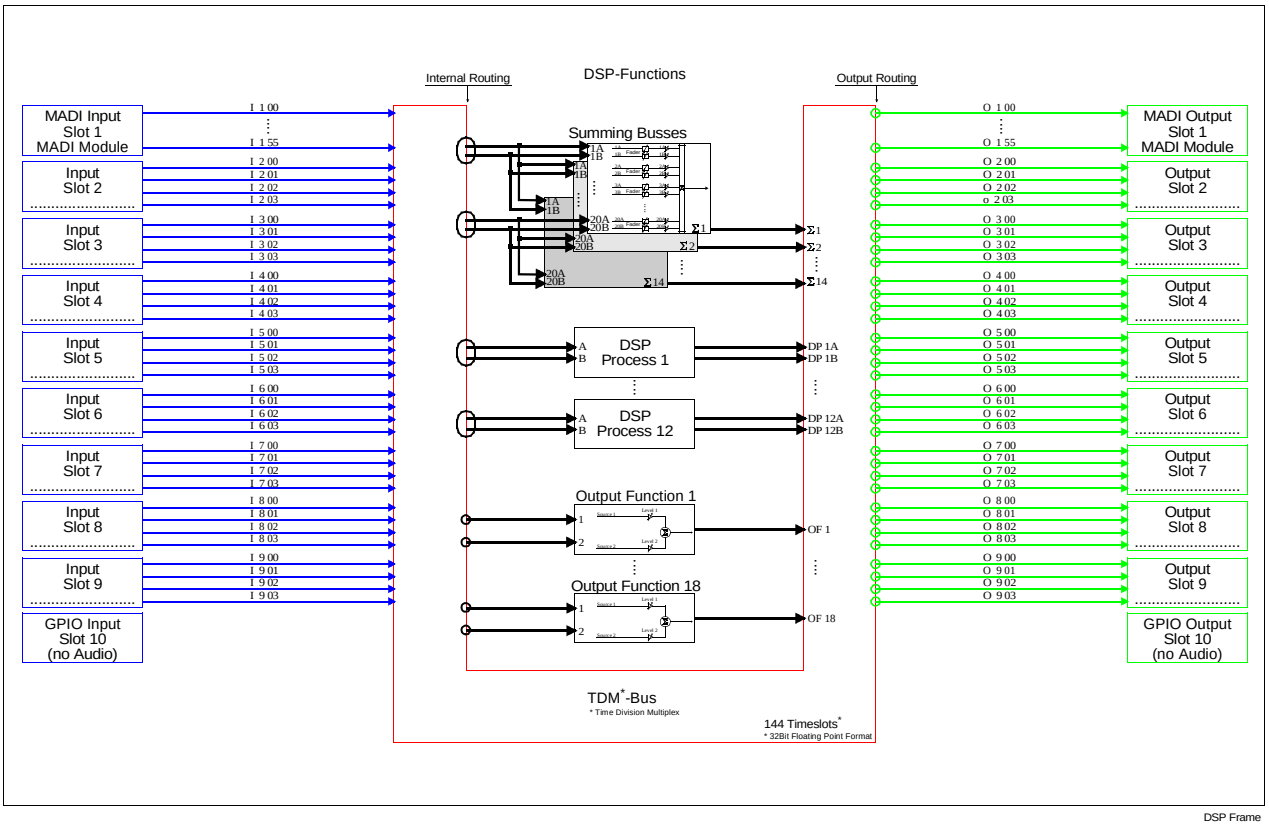
The number of the available Inputs/Outputs, Summing Busses, Output Functions and DSP Processes are shown in Table 3 below.

DSP Frame Type:	RM330-053	RM330-052	RM330-051	RM330-042	RM330-041	
Number of Audioslots	24	16	8	16	8	each with 4 inputs and/or 4 outputs
Number of MADI-Slots	3	2	1	0	0	each with 56 inputs and/ or 56 outputs
Max. number of inputs (1)	264	176	88	64	32	you can pair 2 consecutive channels to 1 stereo channel
Number of DSPs	12	8	4	8	4	ADSP21065-240MHz
Number of Summing Busses (mono)	42	28	14	28	14	each with 20 mono- or 20 stereo inputs
Number of Output Functions (mono)	54	36	18	36	18	each with 2 inputs (mono)
Number of DSP-Processes (stereo or mono)	36	24	12	24	12	each DSP-Process can be used as stereo or mono
Max. number of outputs (1)	264	176	88	64	32	you can pair 2 consecutive channels to 1 stereo channel

Table 3: Overview of DSP Frames

In the following figures (Figure 34 to Figure 38 you will find a scheme of the internal structure of the 5 different DSP Frames.

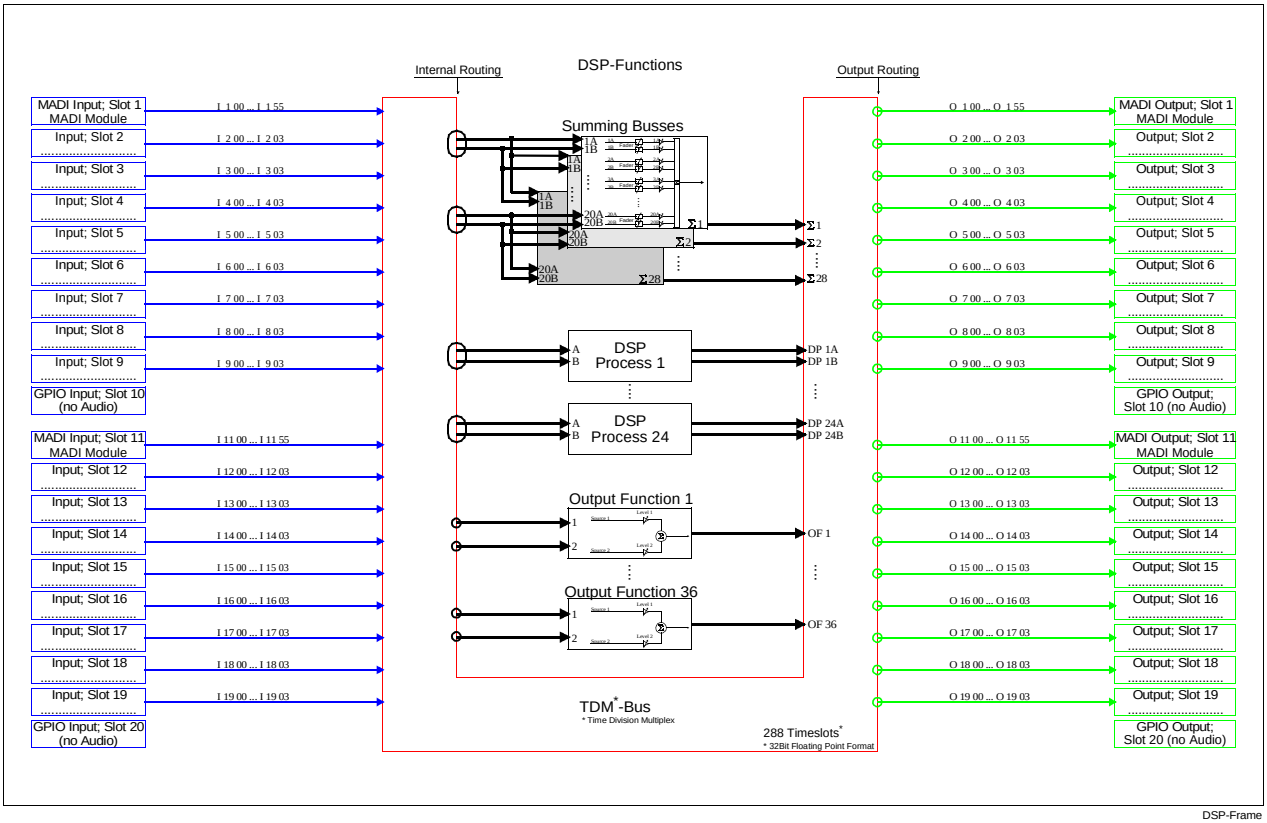
Also, you will see the numbering of the physical inputs and outputs. The number of the inputs and outputs correspond with the slot number and the channel number of each slot. The channel number starts always with zero. The prefix of an input is “I” and of an output “O”.



Names Input-/ Output-Channel: I 1 02

I=Input **O**=Output — Number of Slot — Number of Channel

Figure 34: 10 Slot DSP Frame RM330-051



Names Input-/ Output-Channel:

I 14 02

I = Input, **O** = Output — Number of Slot — Number of Channel

Figure 35: 20 Slot DSP Frame RM330-052

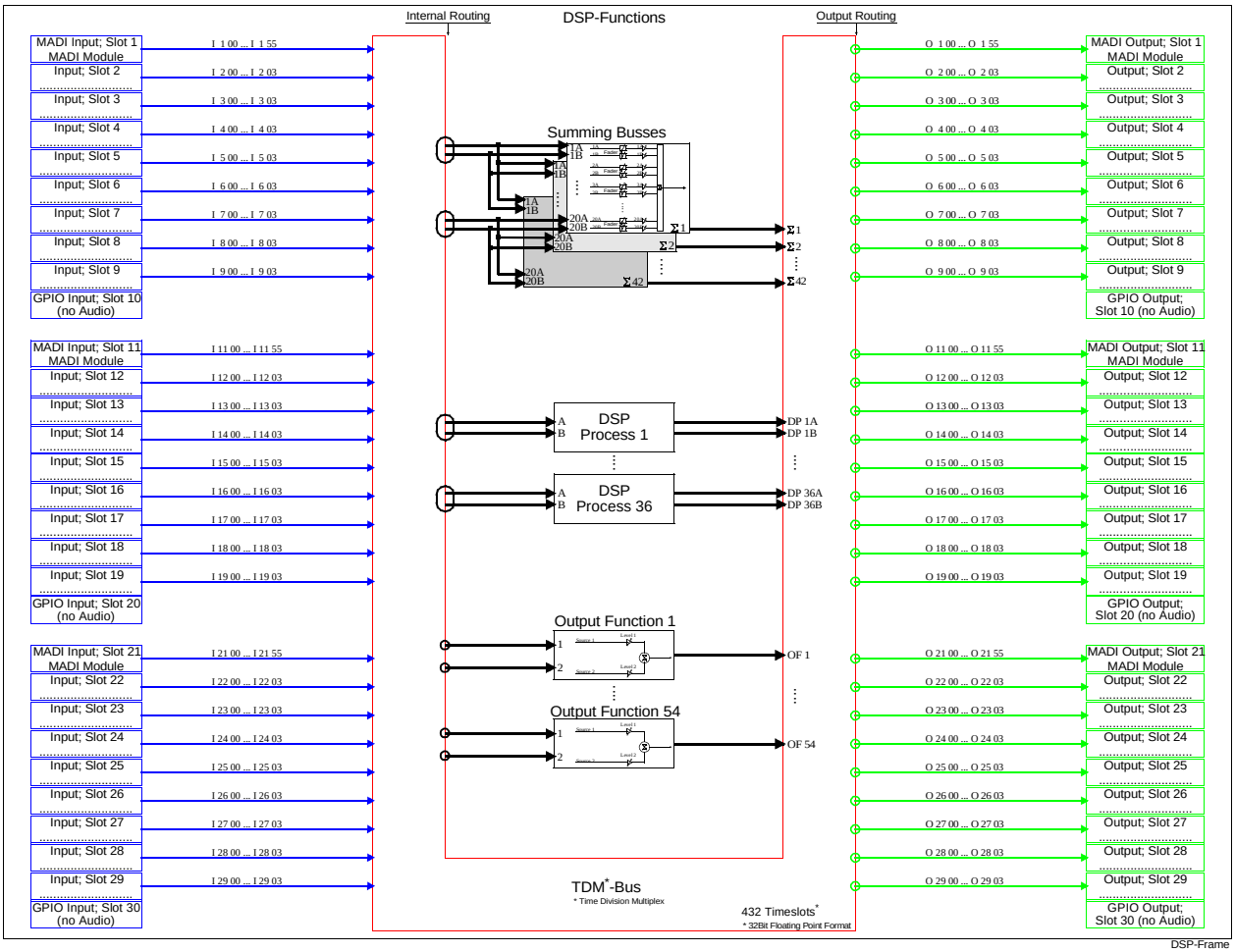
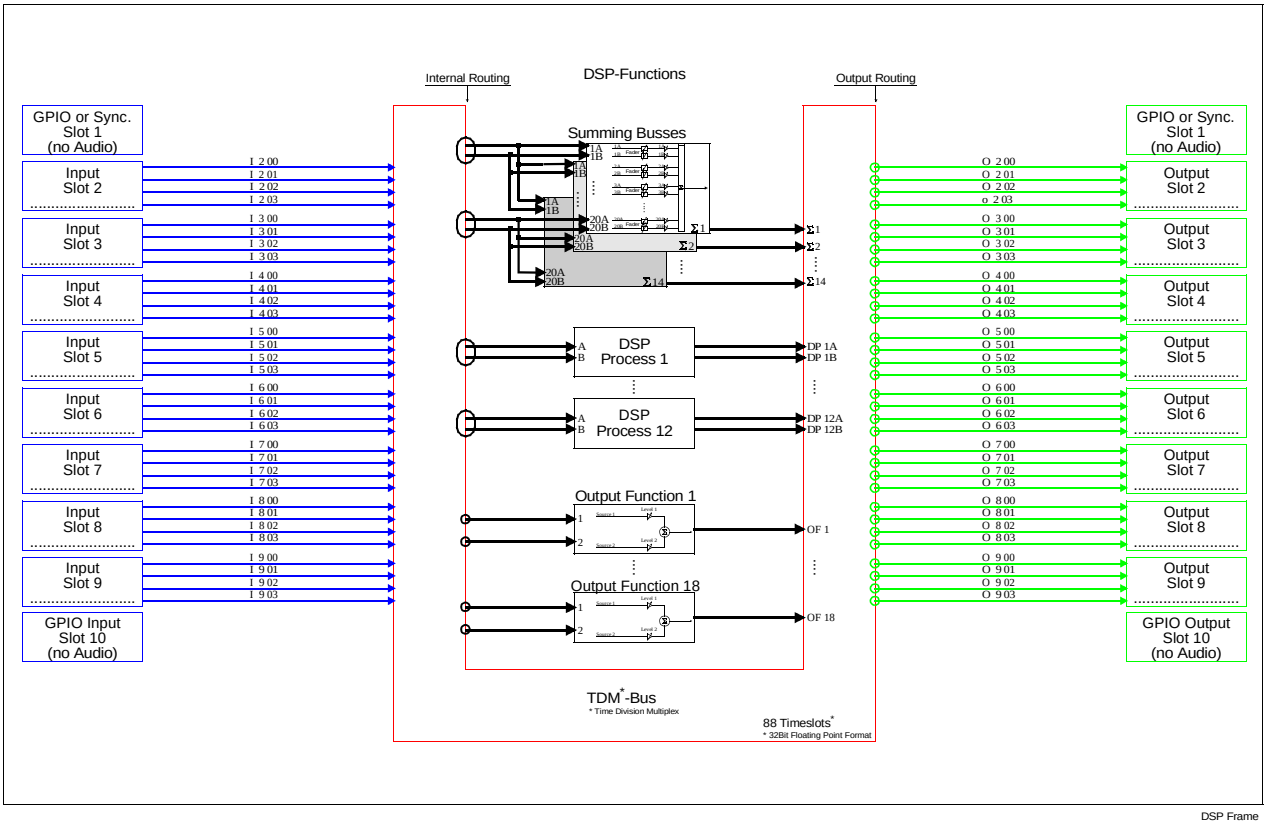


Figure 36: 30 Slot DSP Frame RM330-053



Names Input-/ Output-Channel:

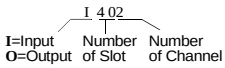
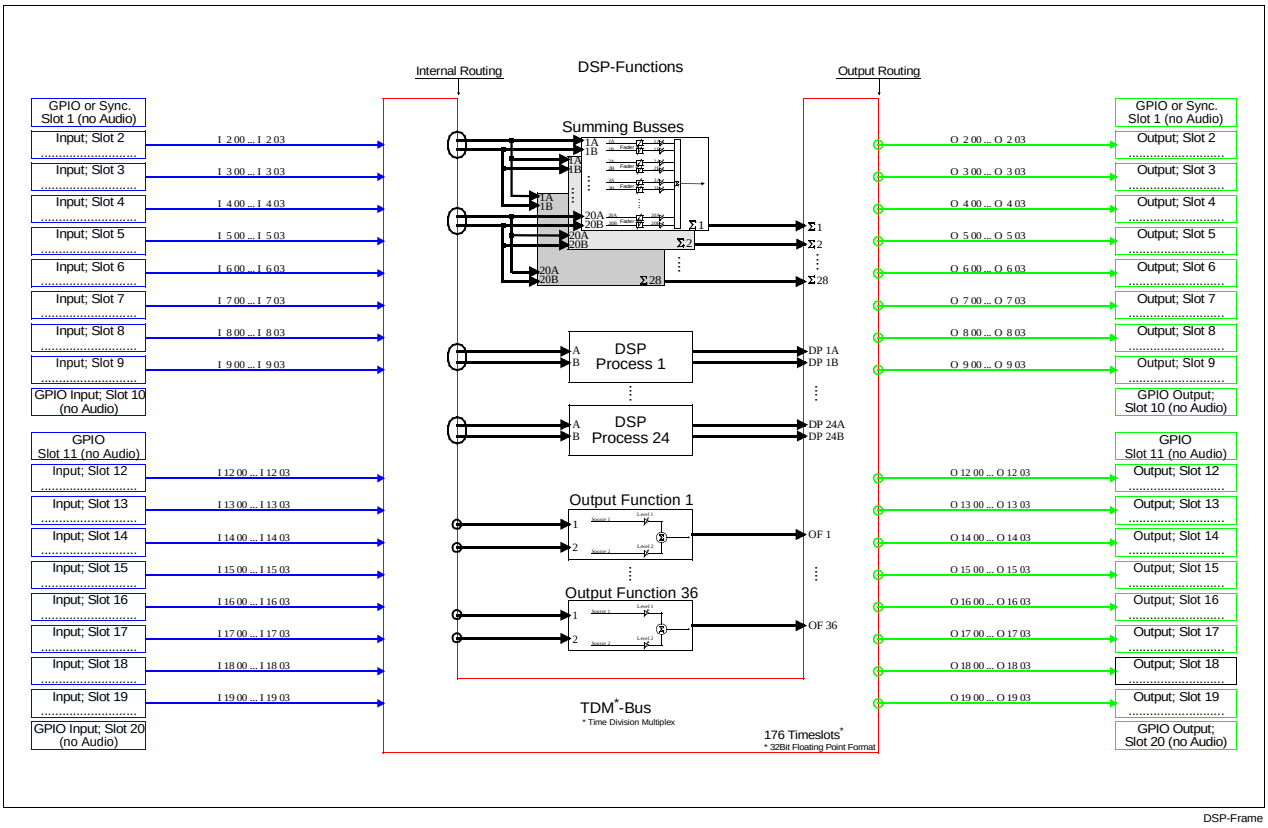


Figure 37: 10 Slot DSP Frame RM330-041



DSP-Frame

Names Input-/ Output-Channel:

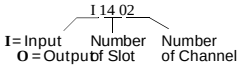


Figure 38: 20 Slot DSP Frame RM330-042

3.2.0.1 Summing Busses

We can use our 14, 28 or 42 Summing Busses for different applications: PFL, Program Bus, Aux Bus and Conference Matrix applications (mix minus).

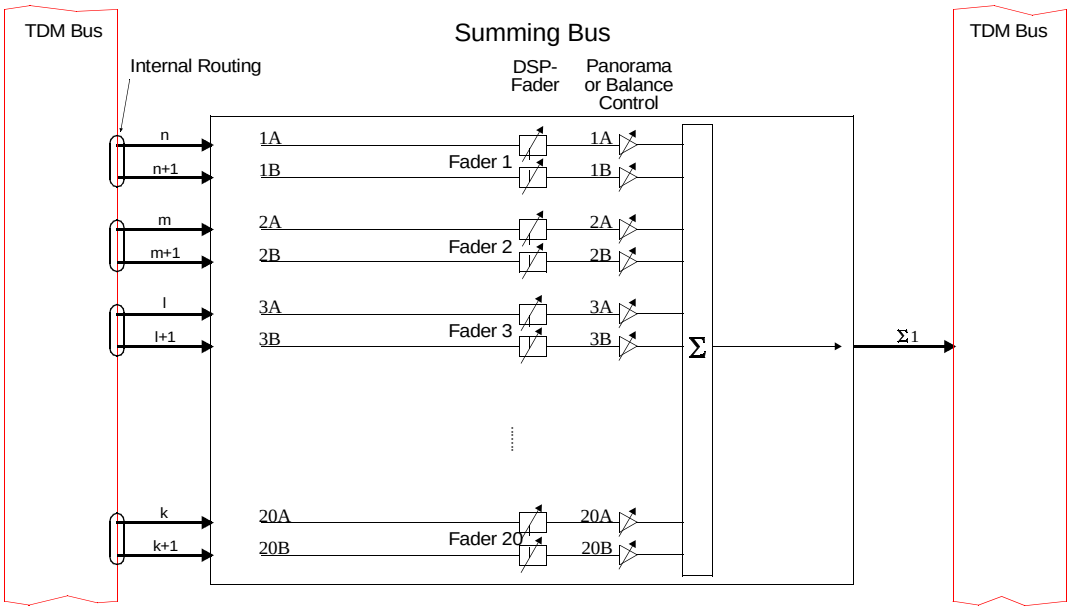


Figure 39: Summing Bus

Each of the max. 42 Summing Busses consists of a sum with 20 stereo inputs (1A, 1B to 20A, 20B). The B input is always the consecutive input on the TDM bus to the A input - it is a stereo routing to the fader inputs. When the fader is a mono fader, then the B input of the fader is not used.

When the fader is used as a mono fader then the panorama is processed, when the fader is a stereo fader, the balance is processed after the fader. To build a stereo sum (Program Bus or stereo Aux Bus), the system takes 2 consecutive summing busses.

Important Note: All 42 Summing Busses are using the same input routing, the same fader values and the same Panorama/Balance values.

Definition.exe

The configuration of the Summing Busses are in the `Configuration/System Configuration window` under the node: `Mixing Functions` and the node: `Conference Matrix`.

In the node: `Mixing Functions` you can select the number of the used Program Busses, Stereo Aux Busses and Mono Aux Busses.

Definition.exe

You can view the actual resources of your Summing Bus consumption on the blue bar “Resources Mixing Functions” in the window `Configuration/System`.

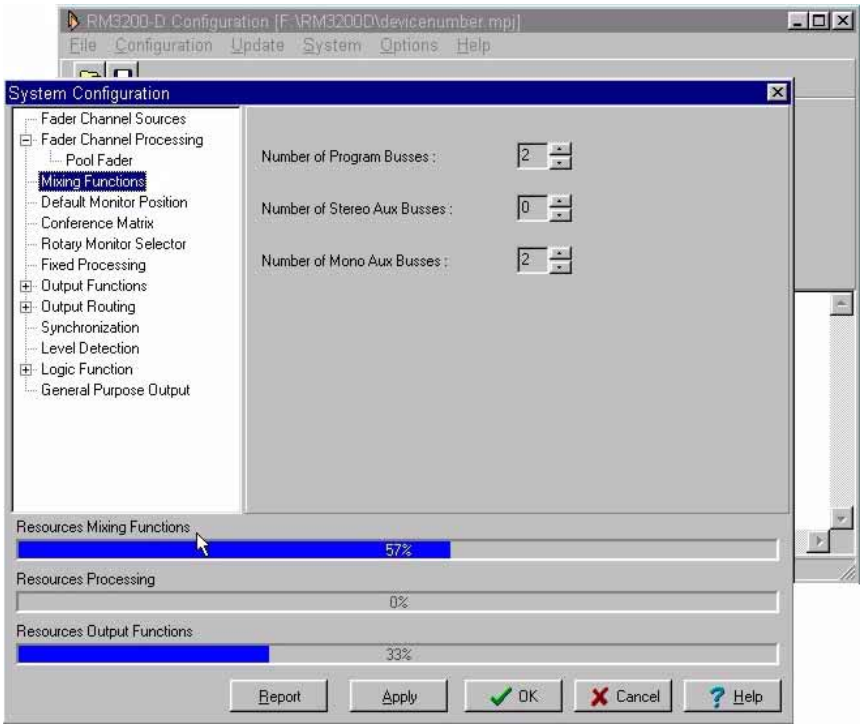


Figure 40: Window “Configuration/System” in the Configuration Software

3.2.0.2 PFL – Pre Fader Listening Bus (Cue Bus)

The PFL Bus always uses the first two Summing Busses (this is fixed in the system software) – this means that PFL is stereo, but when you want to use it as mono, you must use one Output Function (OF) and sum the left and right PFL channels to a mono PFL channel.

Definition.exe

Also, when configuring the PFL Function as non mixed (`Configuration/Console/Fader Module/Function PFL`) – the system always uses these Summing Busses.

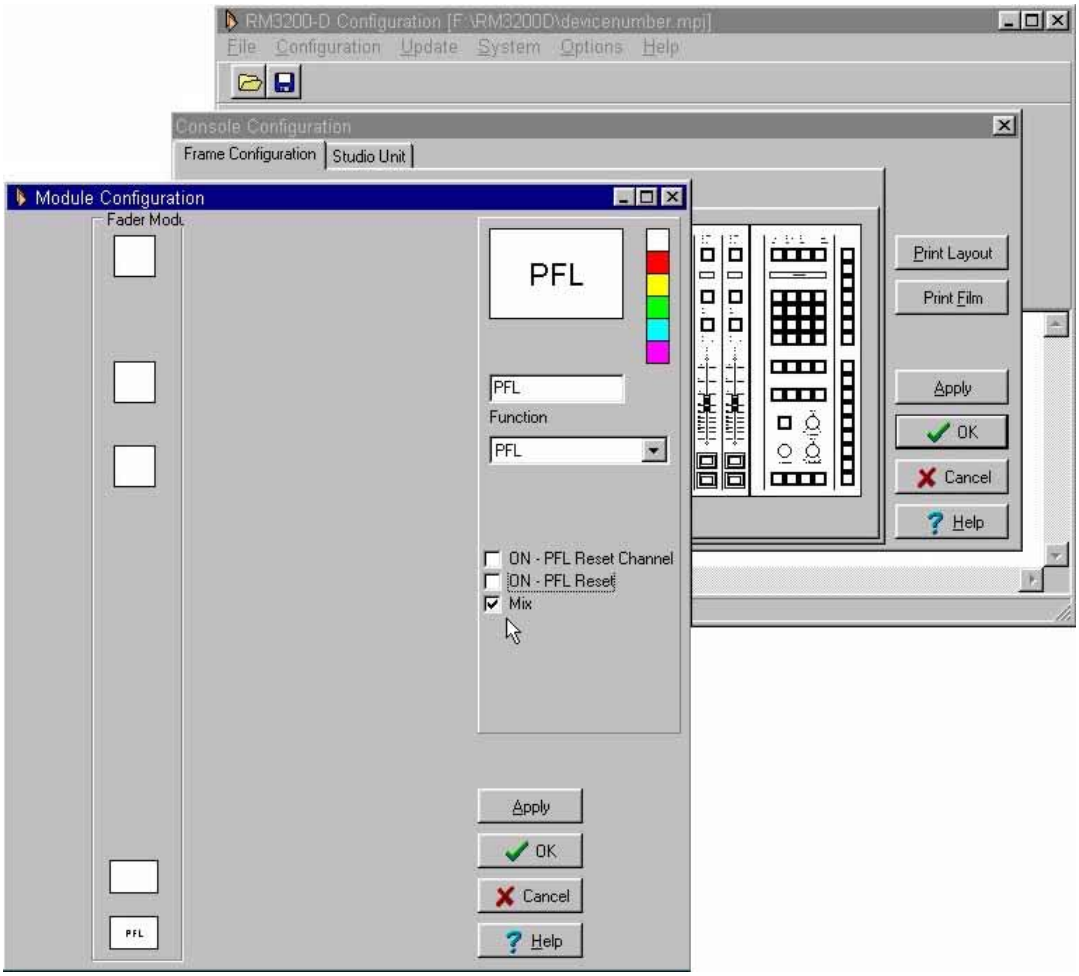


Figure 41:

A Reset Function for the PFL is available in two ways:

1. The PFL resets when the fader in this channel goes from Off to On.
2. The PFL resets when any of the channel faders goes from Off to On.

The configuration for the PFL modes are in the Console Configuration, PFL Button Configuration on the Fader Module.

3.2.0.3 Program Bus

Program Busses are always stereo, after fader and after panorama (when mono input) or balance (when stereo input) process. You need two Summing Busses to build one Program Bus (`Configuration/System` node: `Mixing Functions`).

Definition.exe

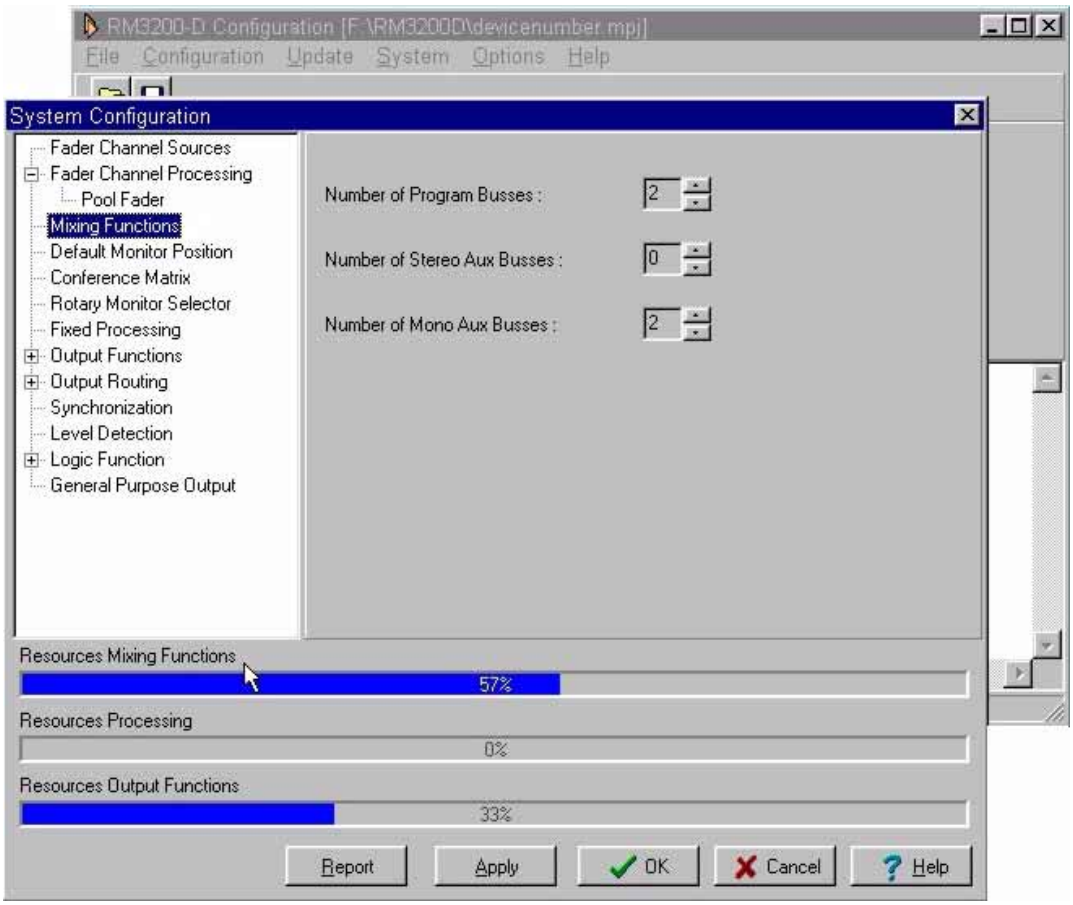


Figure 42:

The first two Program Busses are displayed with green LEDs on the Fader Module when the Bus is selected to the input channel.

The maximum number of Program Busses is 20.

3.2.0.4 Aux Bus

Definition.exe

Aux Busses are mono or stereo, depending on the configuration (`Configuration/System` node: `Mixing Functions`).

Aux Busses can be switched to four different operating modes:

1. ***Pre Fader***
2. ***Post Fader*** (before the panorama or balance process)
3. ***Program*** Bus (this means after panorama or balance) process – here you can build mono Program Busses
4. ***Pre Fader Switch*** – this is a special function, the input goes only to the pre fader aux when the fader is closed

You can switch the aux mode on the control desk when you press the “Aux n Button” twice, i.e. the second menu level. In the first menu level, you can change the gain from the channel to the Aux Bus.

Definition.exe

The configuration of these buttons is in the `Configuration/Console/Main Module/Systemfunction/Aux`. Also, you can configure a Function Select Button (to save buttons when you have a lot of Aux Busses) in the `Configuration/Console/Main Module/Systemfunction/Select Function`. (See “4.2 Configuration of the control desk” on page 89.)

The maximum number of Aux Busses is 31 (all mono plus all stereo), 20 for Stereo Aux Busses and 31 for Mono Aux Busses.

3.2.0.5 Conference Matrix (Mix Minus, n-1)

With this function it is very easy to build Back Signal Busses for mix minus functions or a conference matrix. For every Back Signal Bus you need one (mono) or two (stereo) Summing Busses. The number of Back Signal Busses is 10 (mono or stereo) – you can build a conference with up to 10 parties, each channel can be mono or stereo.

Definition.exe

`(Configuration/System` node: `Conference Matrix)`

3.2.1 DSP Process

The ***DSP Process*** is a “Box” consisting of one or more DSP Functions. We use the DSP Process in two ways, the **Input Processing** and the **Fixed Processing**.

3.2.1.1 Input Processing

The ***Input Processing*** is used to process an input before a fader.

Important Note: The parameters of the Input Process can be changed *only* on the control desk or with the PC DSP control software ACCESS.EXE.

The available DSP functions are:

- Equalizer
- Compressor
- Expander
- Limiter
- Noise Gate
- Deesser

3.2.1.2 Fixed Processing

The ***Fixed Processing*** is used to process for example a Limiter after a Summing Bus.

Important Note: The parameters of the Fixed Processing can be changed *only* with the PC software DEFINITION.EXE or ACCESS.EXE.

The available DSP Functions are:

- Equalizer
- Compressor
- Expander
- Limiter

- Noise Gate
- Deesser
- Transient Limiter (T-Limiter)
- Stereo Enhancer
- Automatic Gain Control (AGC)
- Multiband Compressor/Limiter
- Clipper

The DSP Functions AGC, Stereo Enhancer, Multiband Compressor/Limiter and Clipper are only available as Fixed Processing Functions, and you must buy a special license to use them.

Also, some special DSP Functions are available:

- Level Detector (Input: Audio Signal, Output: Logical Signal)
- Sine Generator

3.2.1.3 DSP Process Description

It is possible to use the DSP Process “Box” mono or stereo.

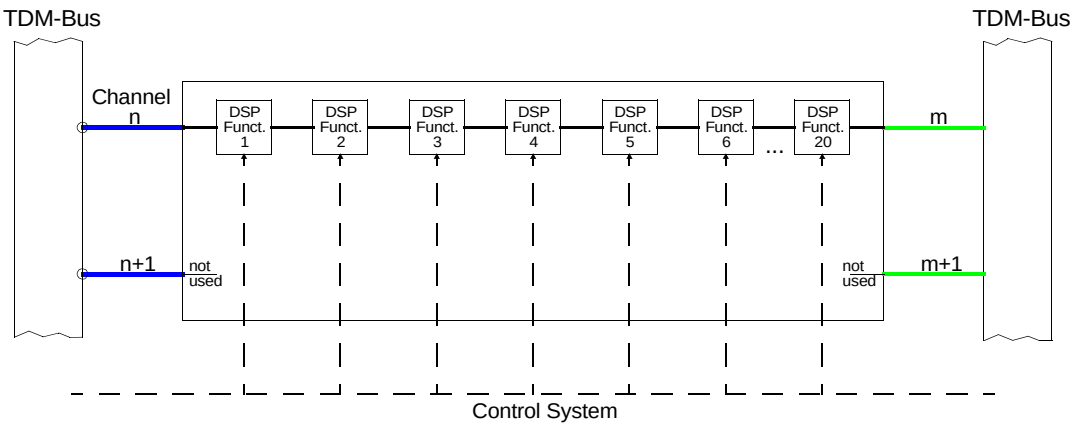


Figure 43: DSP Process Mono

The Input Routing from the TDM Bus into the DSP Process is *always* in stereo (or 2 channel mode) - in the case of a mono DSP Process, the second input B is not used.

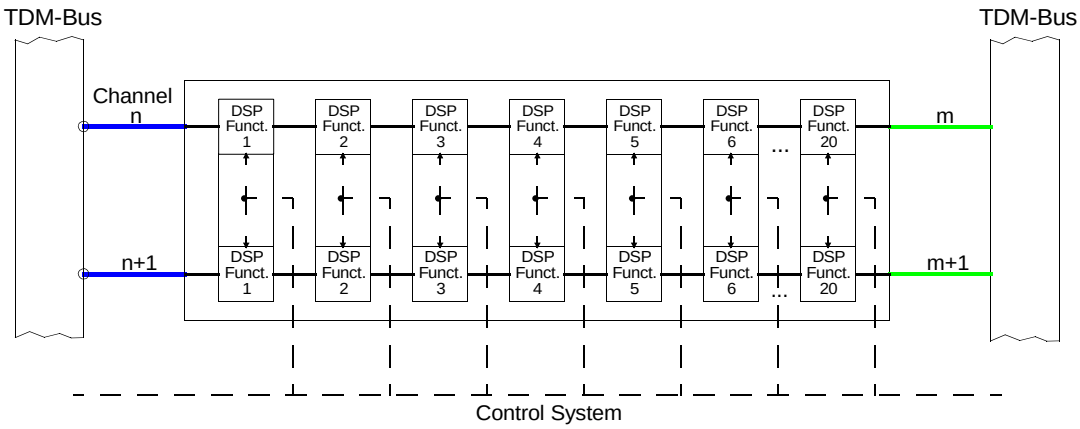


Figure 44: DSP Process Stereo

When used as stereo DSP Process, the parameters (settings) of the functions are the same for the left and the right channel processings. Also, dynamic functions like a Limiter are automatically stereo coupled with the other channel in this DSP Process.

A maximum number of 20 DSP functions is possible within one DSP Process “Box”.

An empty DSP Process works as “bypass” function.

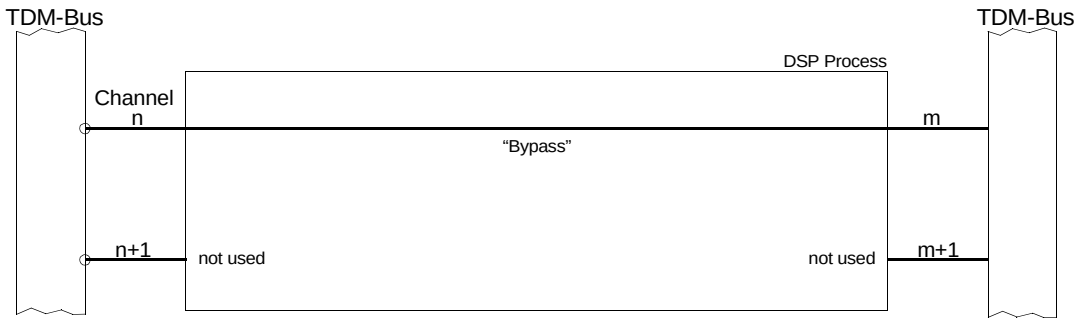


Figure 45: Bypass DSP Process

When more than one DSP Functions of the same type is used, the DSP Control System counts the internal ***Device Number*** (starting with 1) automatically.

Important Note: When using more than one DSP Function of the same type (normally only the Equalizer Function), you must select the right ***Device Number*** on the Push Button Configuration of the control desk.

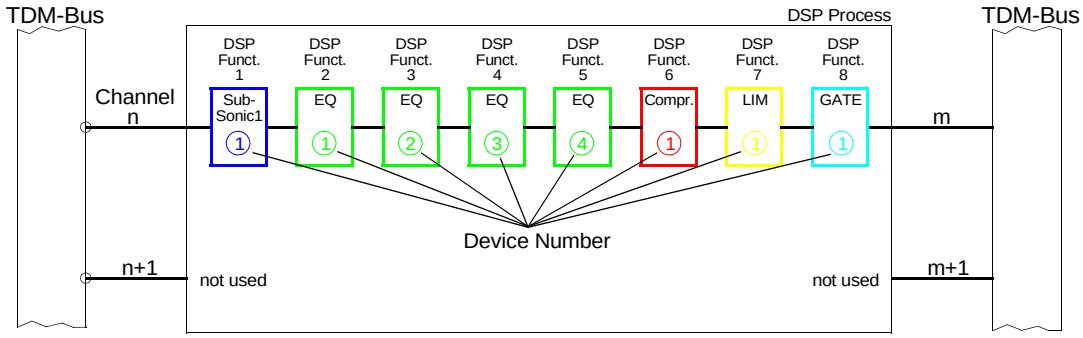


Figure 46: Mono DSP Process with several DSP Functions

There are no restrictions in the order of the DSP Functions, see example below.

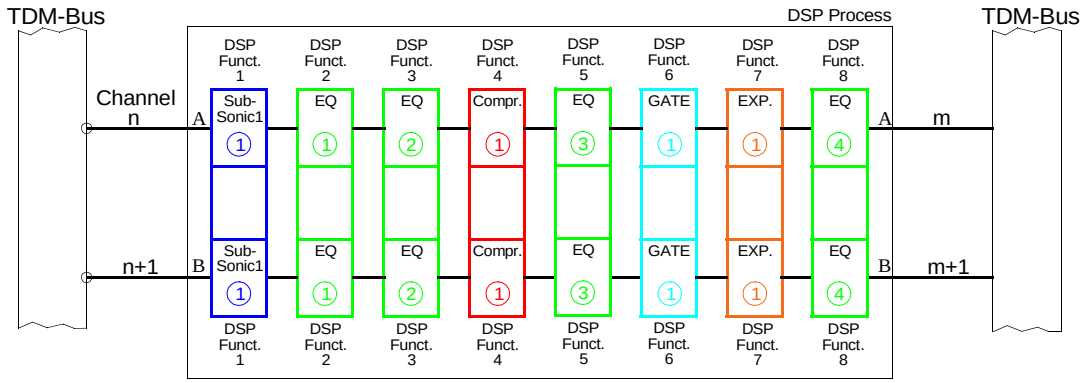


Figure 47: Stereo DSP Process with several DSP Functions, EQs 1 to n

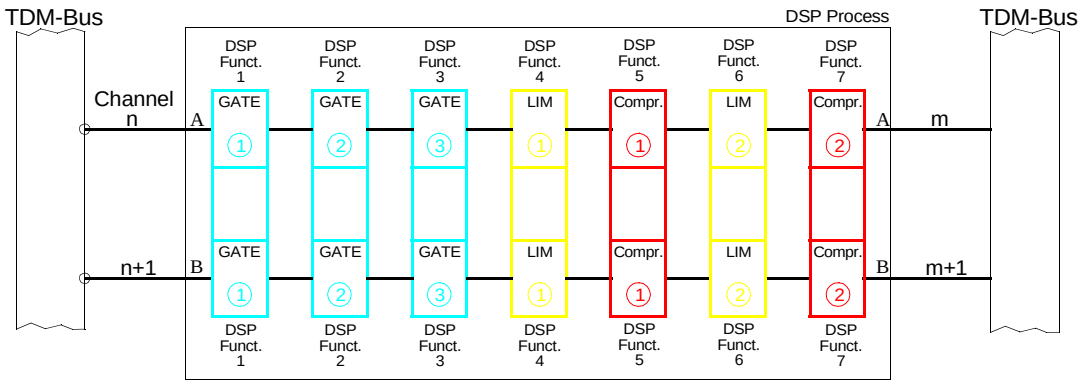


Figure 48: Stereo DSP Process with several DSP Functions, Dynamics 1 to n

Important Note: When using the DSP Process for a Fixed Processing, then the DSP Process can be used only mono. It is possible to let up arrange a stereo coupling of the two paired devices in the Configuration Software.

Only the Input Processing of a fader channel can be used as stereo (or mono).

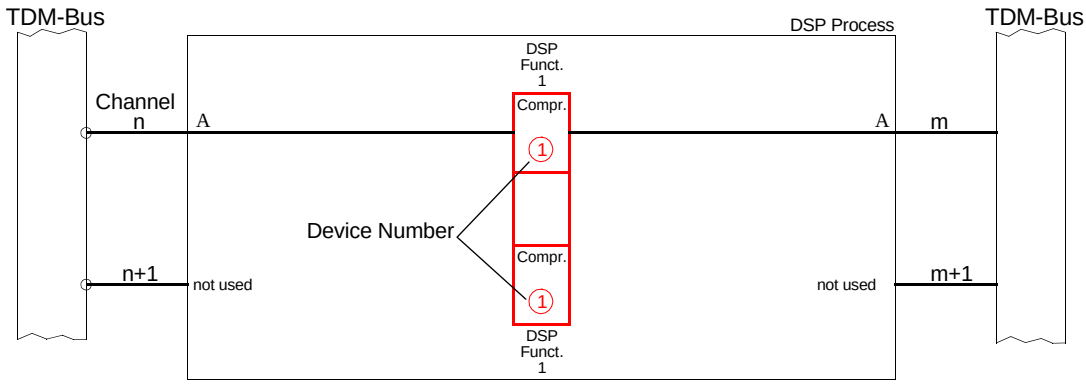


Figure 49: Mono DSP Process with one DSP Function

3.2.2 Output Function

The key to solve special problems in a broadcast mixing console or other special application is to use the ***Output Function***.

Each of the ***Output Function*** consists of **two inputs** (`source 1` and `source2`) from the TDM Bus **summed together with** `level 1` (for the signal from `source 1`) and `level 2` (for the signal from `source 2`) to the output of the Output Function.

The ***Output Function*** can have *different* sources and levels depending on ***logical conditions***.

The input sources `source 1` and `source 2` can be any audio signal available to the TDM Bus.

The levels `level 1` and `level 2` can be any gain value between -100dB and +20dB in steps of 1dB or “off” (–∞).

It is also possible to connect a ***potentiometer*** to the level controls of an Output Function (see **“Output Function Example PFL/Monitor” on page 62**).

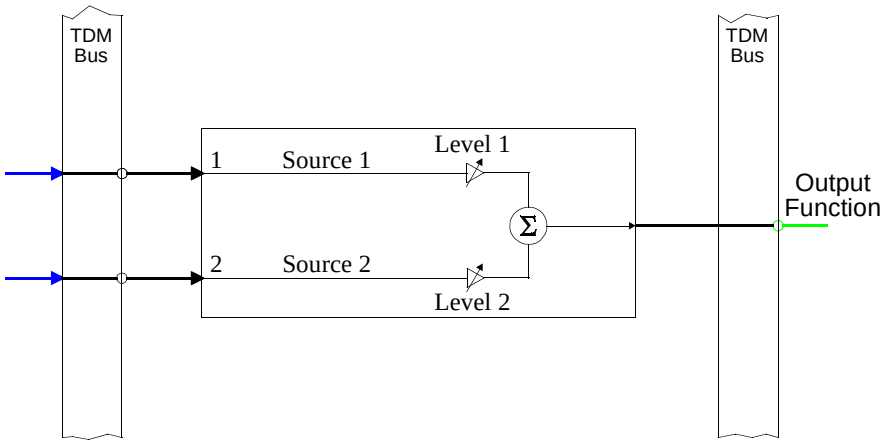


Figure 50: Output Function

Important Note: ***Output Functions*** are the *interconnection* between the control logic of the system that can be configured by the user and the audio signals in the system.

A ***logical condition*** can be a push button of the RM3200D, a logical control input GPI of the GPIO Module, a Fader Start Signal or some other special logical events.

3.2.2.1 Condition List

The different logical events are listed, to be found in the ***condition list*** of the Output Function.

Important Note: On the top of the ***condition list*** is always the logical condition **None**. None means that **all** *logical conditions in the condition list are* **not true**.

Including the first condition line **None**, up to 10 condition lines are possible in the condition list. The condition list is processed from the top to the bottom.

The first *valid* logical condition below the first line None in the condition list has **priority** over the *following* lines in the condition list.

The output of each Output Function is available on the TDM Bus. To understand the principle of the Output Function, study the following examples carefully.

3.2.2.2 Output Function Example Talk into Aux Bus

In this first simple example, the Output Function is used only as a switch with fix sources and different levels in the condition list.

The application of this example is to speak with the microphone Mic 1, which is available on the TDM Bus into the Aux 1 of the system when a push button on the control desk is pressed.

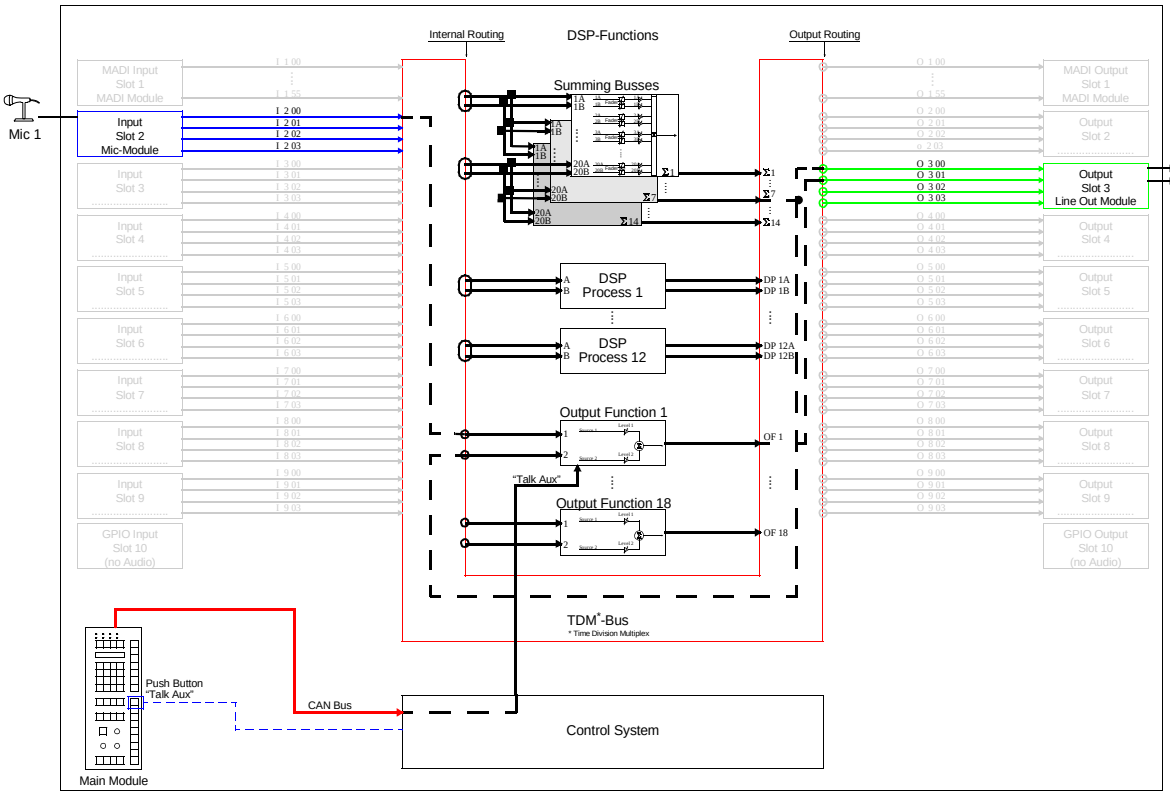


Figure 51: Example Talk Aux

In this example we have two logical conditions:

1. The logical condition **None** means in this example that the push button “Talk Aux” is not pressed.
2. The condition “Talk Aux” push button is pressed - this means the condition “Talk Aux” is true or logical 1

In this simple example, the signal on the output of the Output Function switches simply between the two audio signals under the two logical conditions in the condition list:

1. Logical condition: “Talk Aux” = false = **None** (push button not pressed), source 1: “Mic 1” with level 1: off, source 2: “Aux Bus 1” with level 2: 0 dB
2. Logical condition: “Talk Aux” = true (push button pressed), source 1: “Mic 1” with level 1: 0 dB, source 2: “Aux Bus 1” with level 2: off.

In this example, the routing for `source 1` and `source 2` is not changed under a logical condition, only the `level 1` and `level 2`.

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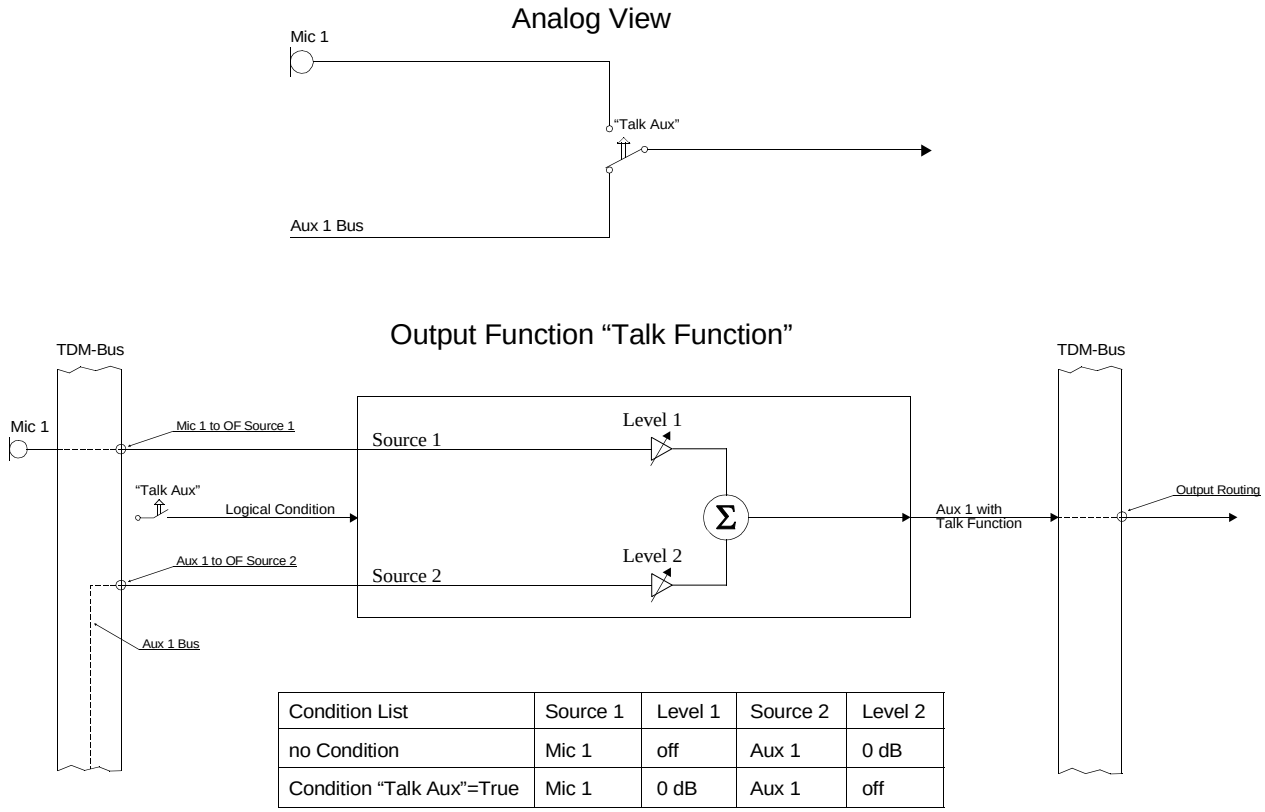

Condition List	Source 1	Level 1	Source 2	Level 2
no Condition	Mic 1	off	Aux 1	0 dB
Condition "Talk Aux"=True	Mic 1	0 dB	Aux 1	off

Figure 52: Analog view of the "Talk Aux" example compared to the solution with the Output Function

On the following pages we give a short description of the configuration of this example. For details see also chapter **"Configuration" on page 80**.

Definition.exe

In the first step, we define the push button on the Main Module RM330-010 in the `Configuration/Console/Module Configuration` with the name "Talk\Aux" and with the Function "Userdefined". After this definition, the push button is now available as logical condition in the control system of the RM3200D.

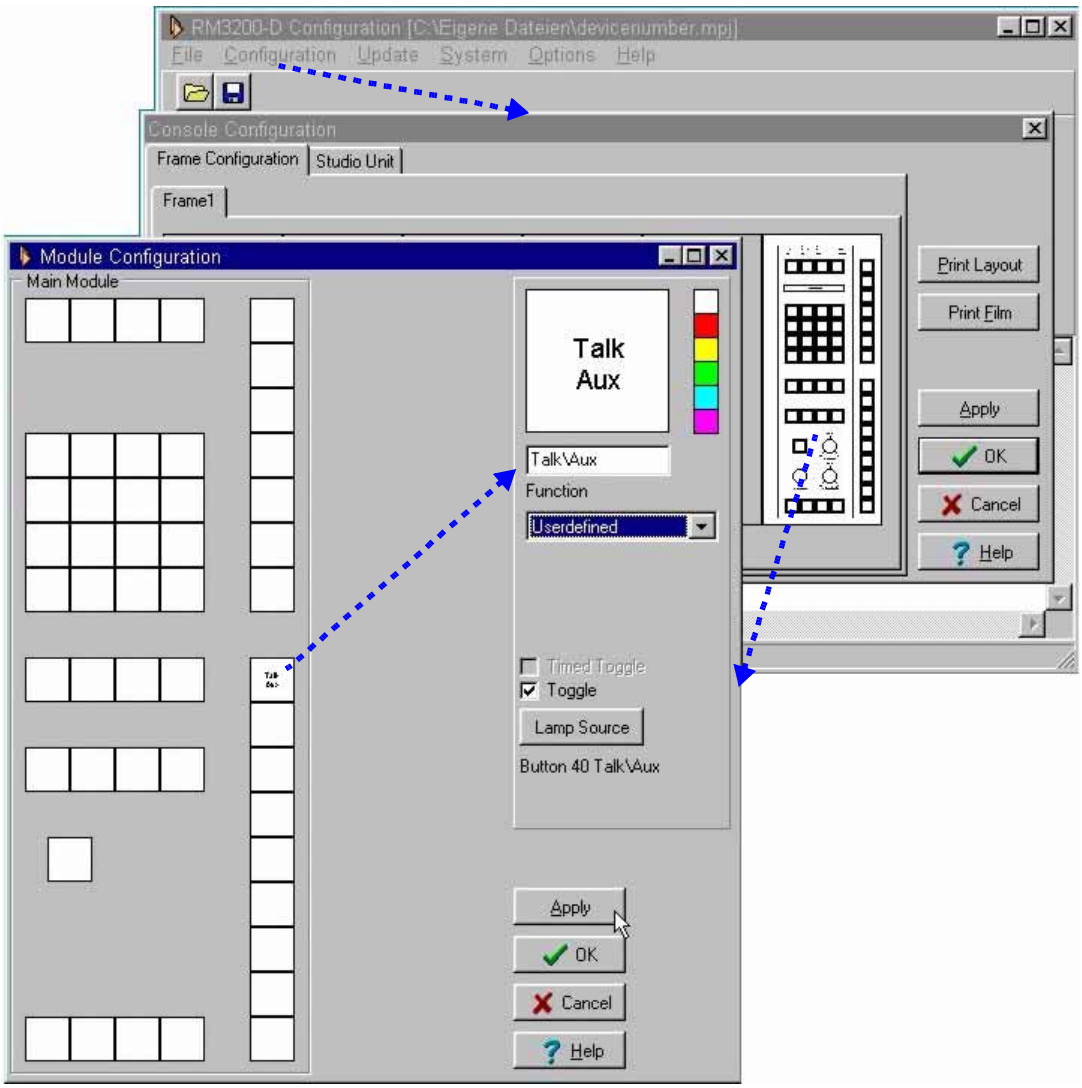

Figure 53: Define Push Button for Output Function „Talk Aux“

Definition.exe

The second step in the configuration process is to build this Output Function in the window `Configuration/System` node: `Output Function` by the following steps:

1. By clicking the right mouse button you must select: `Insert new Output Function`.
2. Mark the new Output Function and give it the name “TalkAux” in the small edit field on the top of the right side (or use the default name).
3. Define the `source 1, level 1, source 2` and `level 2` in the first condition line with the logical condition `None`.

Important Note: Please note that the first line in the condition list has always the logical condition **None** - this logical condition can *not* be changed!

4. Insert the second condition line in the condition list by clicking the right mouse button and selecting `Insert new Condition` (or use the Insert-key on your

PC keyboard).

5. Highlight the second line in the condition list with the mouse, define the logical condition by pressing the button “Select Condition” and pick up the logical condition “TalkAux” in the window “Condition” in the node “Buttons”.

   The `Level/Edge` radiobutton in the window “Condition” must be “On”, this means, when the push button “TalkAux” is pressed, the logical condition “TalkAux” is true or 1.

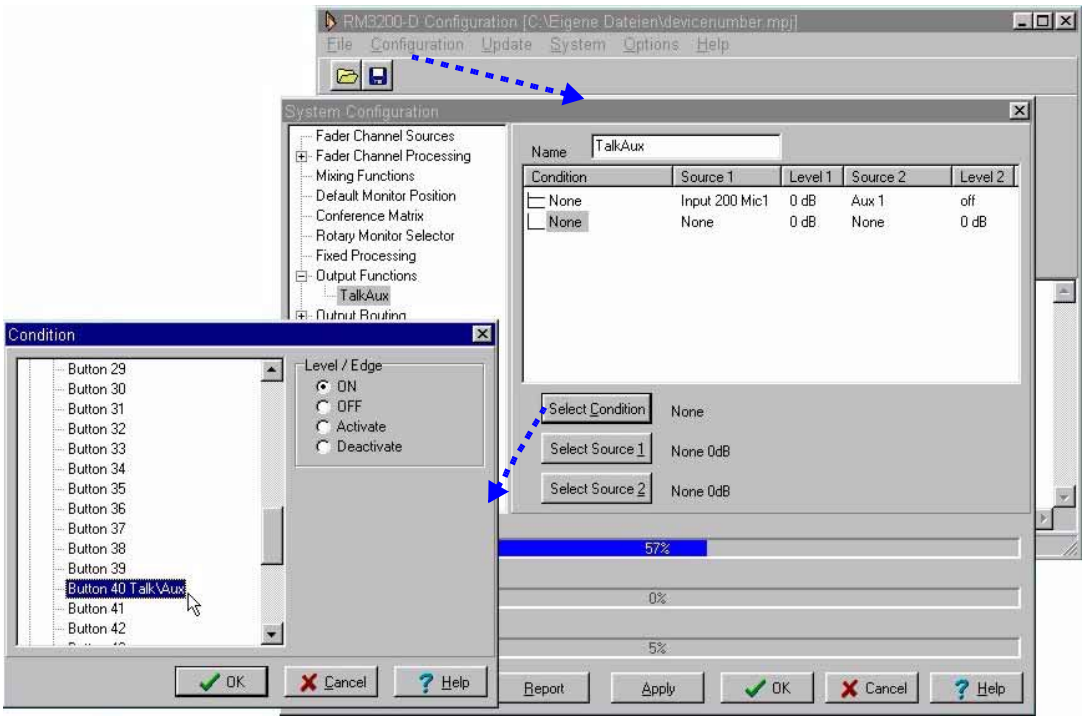

Figure 54: Select condition for the Output Function “Talk Aux”

6. Define the `source 1, level 1, source 2` and `level 2` in the second condition line with the logical condition `“Button 40 Talk\Aux”`.

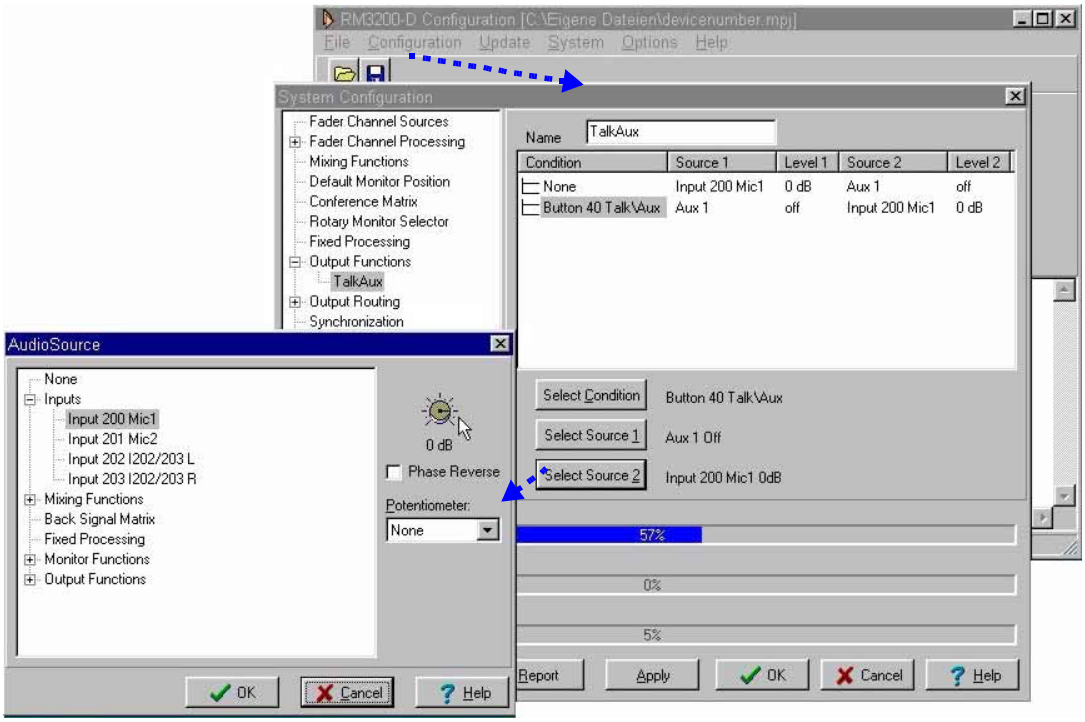

Figure 55: Select sources for the Output Function “Talk Aux” with logical condition “Button 40 Talk\Aux”

After the definition of the Output Function “TalkAux”, the Output Function “TalkAux” is now available as audio signal on the TDM Bus.

Definition.exe

To use the audio signal, you must route this signal in the `Configuration/System` node: `Output Routing` to a physical output or into another DSP Function like a Summing Bus (as fader channel input), Output Function or Fixed DSP Process.

3.2.2.3 Potentiometer

To control the `level 1` and `level 2` of the output functions, you can also connect a potentiometer to the level controls.

Important to understand for the user is, that you connect internal potentiometer values to the level controls of the Output Function.

The internal potentiometer values are coming from a “potentiometer bus” with several input possibilities:

- Potentiometers from the Main Module or from the Studio Units (pre labelled with “Monitor” and “Headphone” but usable for any level control function!)
- Potentiometers built with a push button + the rotary control knob from the Main Module or from the Studio Units
- Analog Control Inputs from the GPIO modules
- Fadervalues from the faders of the system or from faders of another connected RM3200D (via RS232/422) - the ***Global Potentiometers***.

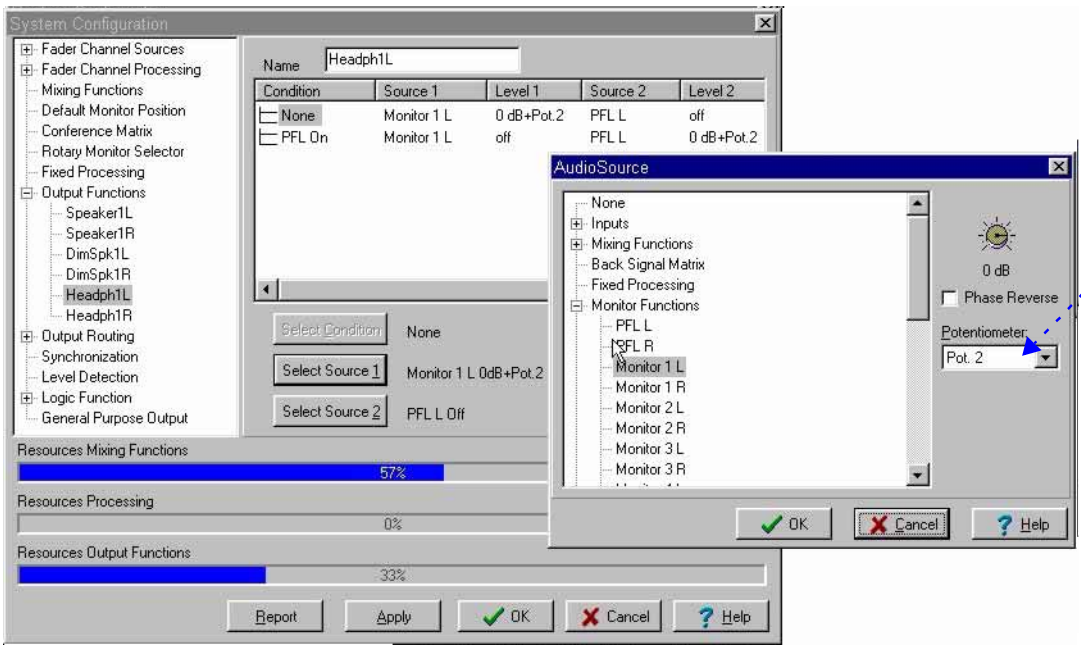

Figure 56: Select a potentiometer for the level control in a condition line

The following potentiometers are available for the Output Functions:

Pot.1 to Pot.20

Pot.	Default Potentiometer Source	Reset Value
1	“Monitor” Potentiometer from the Main Module or Analog Control Input or Rotary Control	off
2	“Headphone” Potentiometer from the Main Module or Analog Control Input or Rotary Control	off
3	“Monitor” Potentiometer from the Studio Unit 1 or Analog Control Input or Rotary Control	off
4	“Headphone” Potentiometer from the Studio Unit 1or Analog Control Input or Rotary Control	off
5	“Monitor” Potentiometer from the Studio Unit 1or Analog Control Input or Rotary Control	off
6	“Headphone” Potentiometer from the Studio Unit 1 or Analog Control Input or Rotary Control	off
7 to 10	Analog Control Input or Rotary Control	off
11 to 20	Analog Control Input or Rotary Control	0dB

Table 4: Potentiometer Sources

Caution: Take care when you configure Pot.1 to Pot.6 to an Analog Control Input or a Rotary Control Potentiometer Function. When the Main Module or the Studio Units is connected to the CAN Bus, the default potentiometer sources are automatically occupied. In this case, do not use these Pots as Analog Control Input or Rotary Control Potentiometer Function either!

Normally, the potentiometers 1 and 2 are used in conjunction with four Output Functions to control the levels of your headphones (left, right) and your studio loudspeakers (left, right), refer to example **“Output Function Example PFL/Monitor” on page 62**.

FAQ: Is there any relationship between the Pots 1 to 6 and the Monitor Busses 1 to 6?

Answer: No, the Pots 1 to 6 are completely independent from the Monitor Busses 1 to 6!

Balance 1L, Balance 1R, Balance 2L, Balance 2R
Used for Balance control of Monitor Loudspeakers.

Global Pot.1 to Global Pot.20
Interconnection between faders and Output Functions, the global potentiometers are available on the RS232/422 on the system and usable in other connected RM3200Ds.

For example, with a Fader of a RM3200D you can control the level of an Output Function (or more Output Functions) in another RM3200D.

MPX Out 1 to MPX Out 10
To control the output level of a conference matrix (the back signal or the mix minus signal).

3.2.2.4 Output Function Example PFL/Monitor

In broadcast self operating control rooms the main monitor loudspeakers are often also used to pre-listen (PFL or cue) the next events. In the case that one of the PFL push buttons is pressed, the PFL signal goes to the right loudspeaker and the normal monitor signal (commonly your on air signal) goes attenuated to the left loudspeakers.

The headphones are simply switched from the Monitor Bus to the PFL Bus.

At least, the loudspeaker signal can be attenuated by some dBs to build a “Dim” function when pressing a push button on the control surface.

We use six output functions for this example.

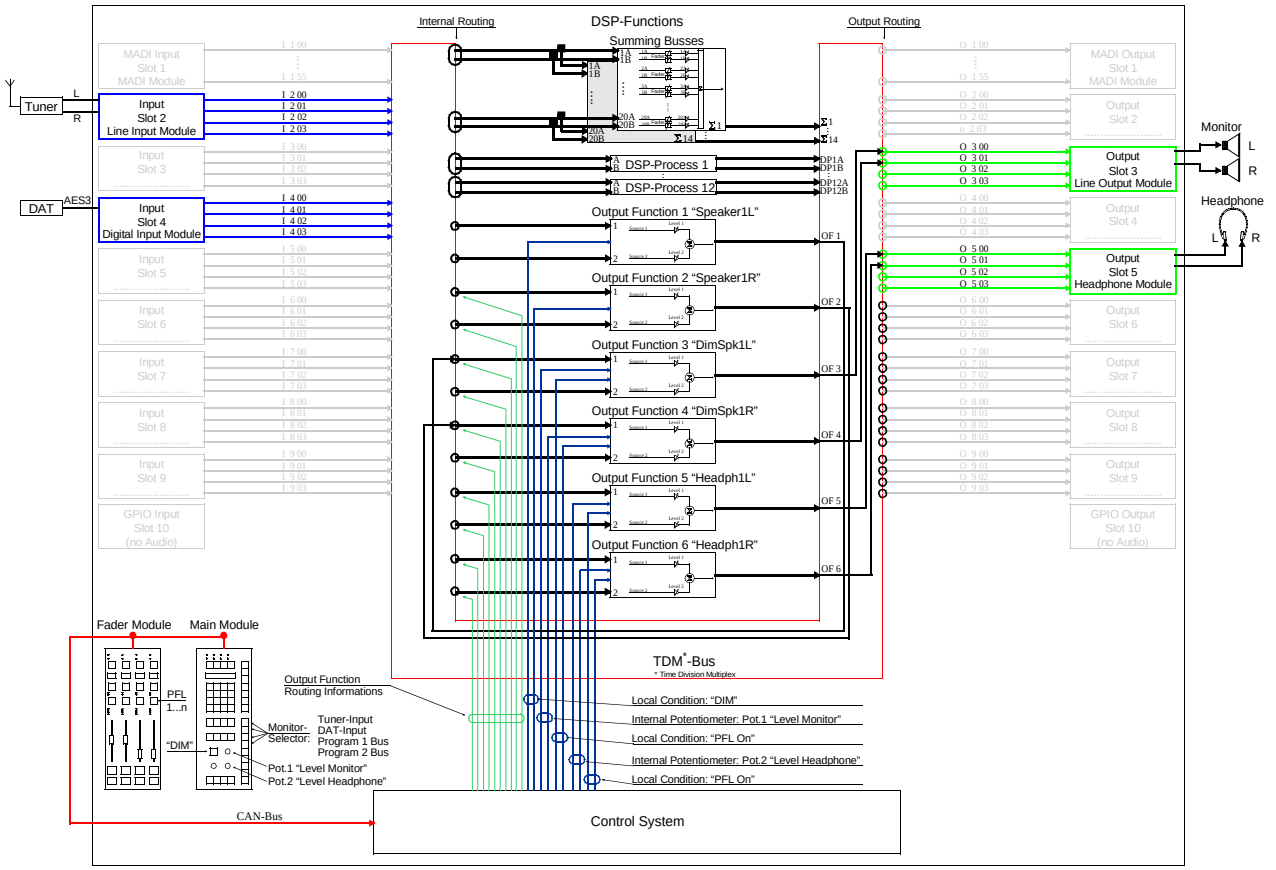

Figure 57: Example PFL/ Monitor

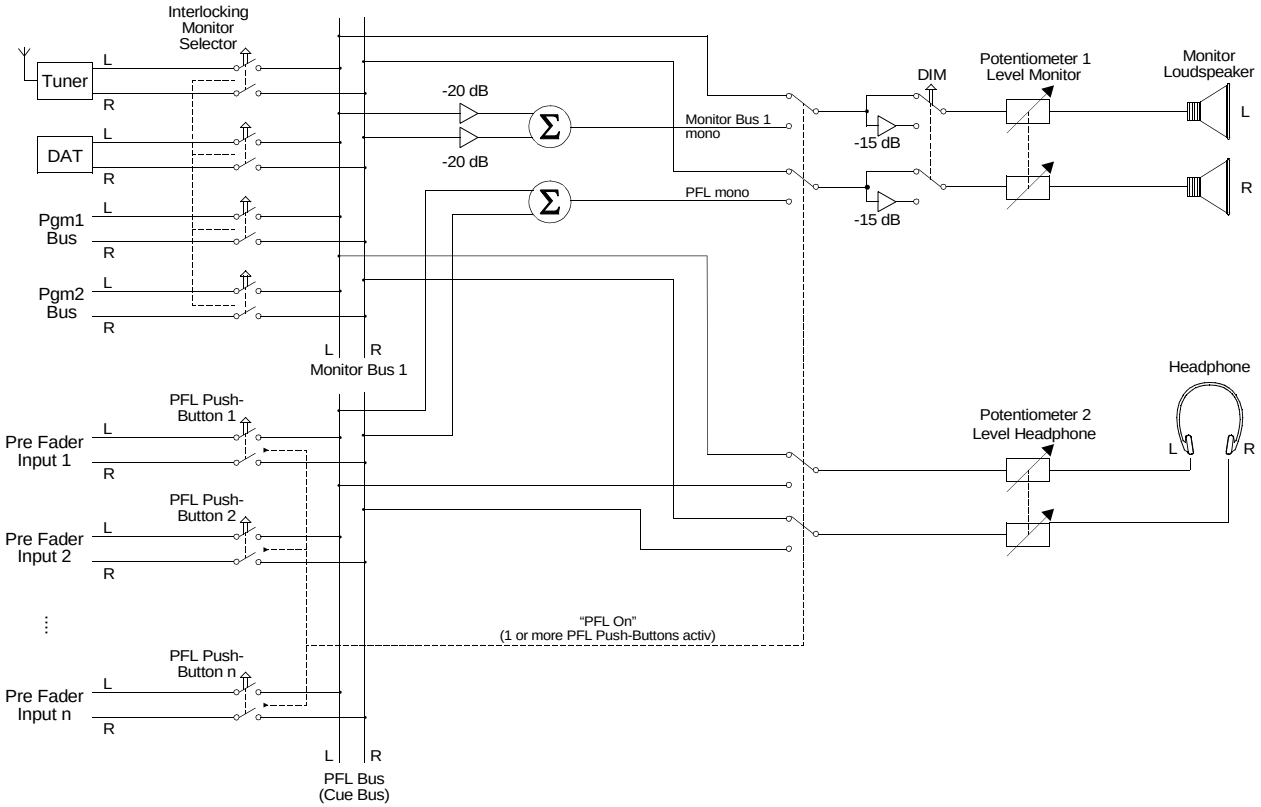

Figure 58: Analog view of the "PFL/ Monitor" example

Output Function 1 “Speaker1L”

Condition list	Source 1	Level 1	Source 2	Level 2
None	Monitor 1 L	0 dB	None	off
PFL on	Monitor 1 L	-20 dB	Monitor 1 R	-20 dB

Table 5: Condition list “Speaker1L”

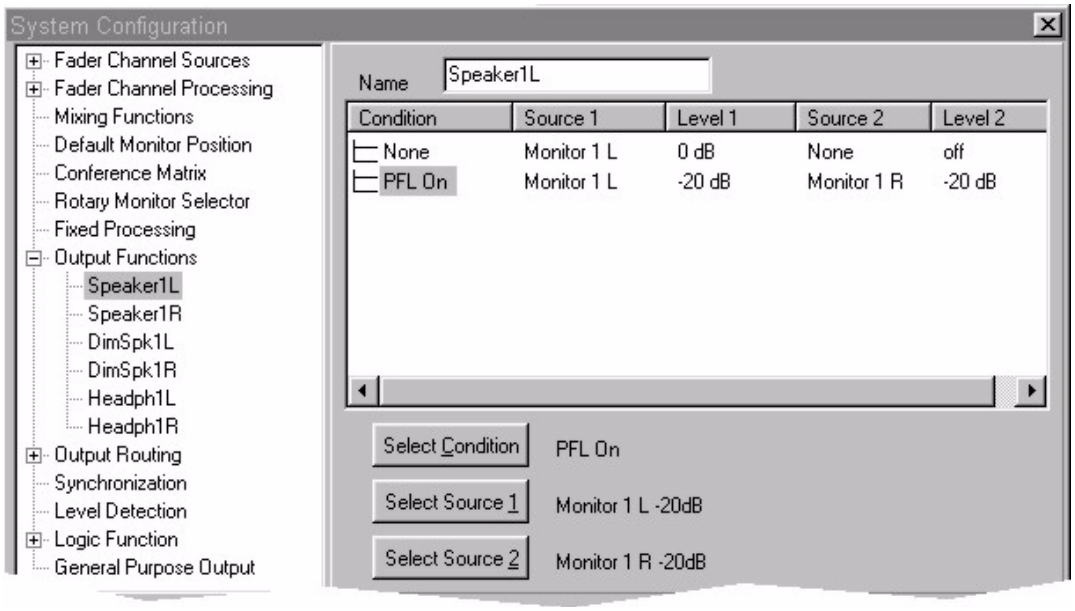

Figure 59: Condition list “Speaker1L”

Output Function 2 “Speaker1R”

Condition list	Source 1	Level 1	Source 2	Level 2
None	Monitor 1 R	0 dB	None	off
PFL on	PFL L	0 dB	PFL R	0 dB

Table 6: Condition list “Speaker1R”

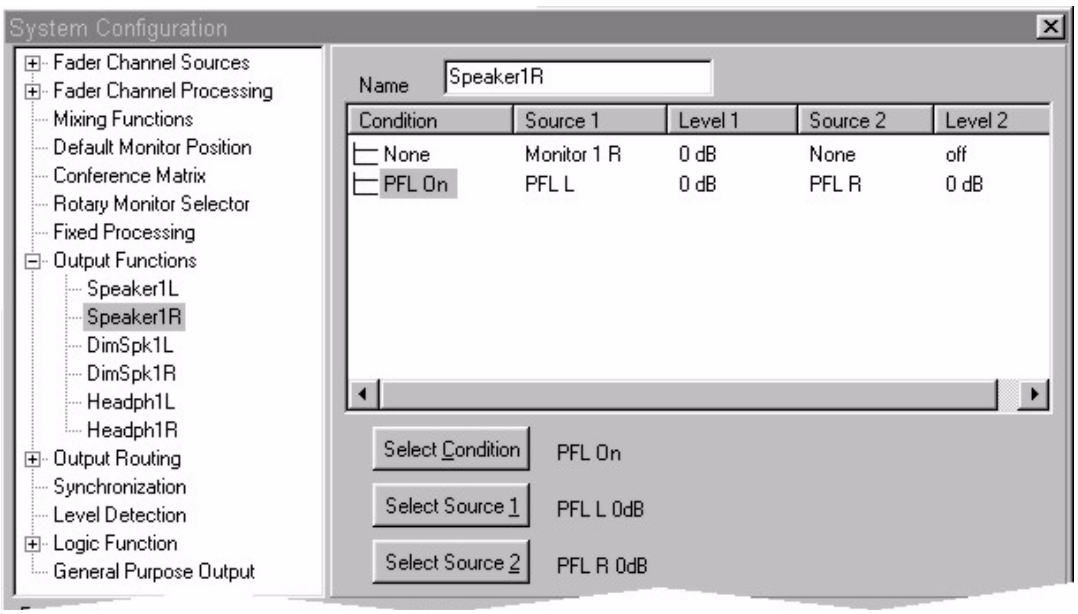

Figure 60: Condition list “Speaker1R”

Output Function 3 “DimSpk1L”

Condition list	Source 1	Level 1	Source 2	Level 2
None	Speaker1L	0 dB + Pot.1	None	off
Button 49 “DIM”	Speaker1L	-15 dB + Pot.1	None	off

Table 7: Condition list “DimSpk1L”

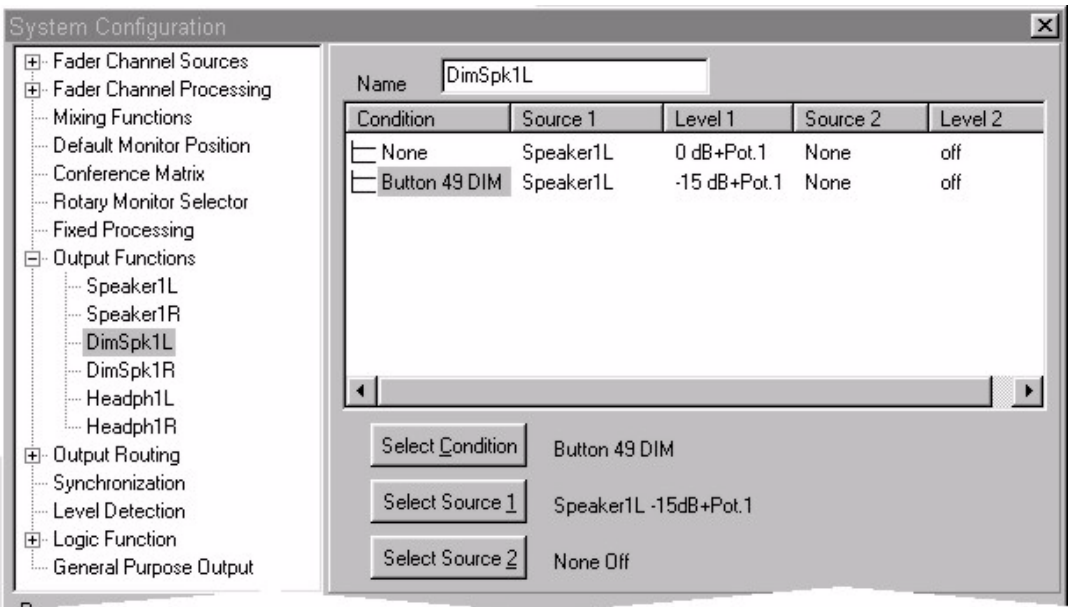

Figure 61: Condition list “DimSpk1L”

Output Function 4 “DimSpk1R”

Condition list	Source 1	Level 1	Source 2	Level 2
None	Speaker1R	0 dB + Pot.1	None	off
Button 49 “DIM”	Speaker1R	-15 dB + Pot.1	None	off

Table 8: Condition list “DimSpk1R”

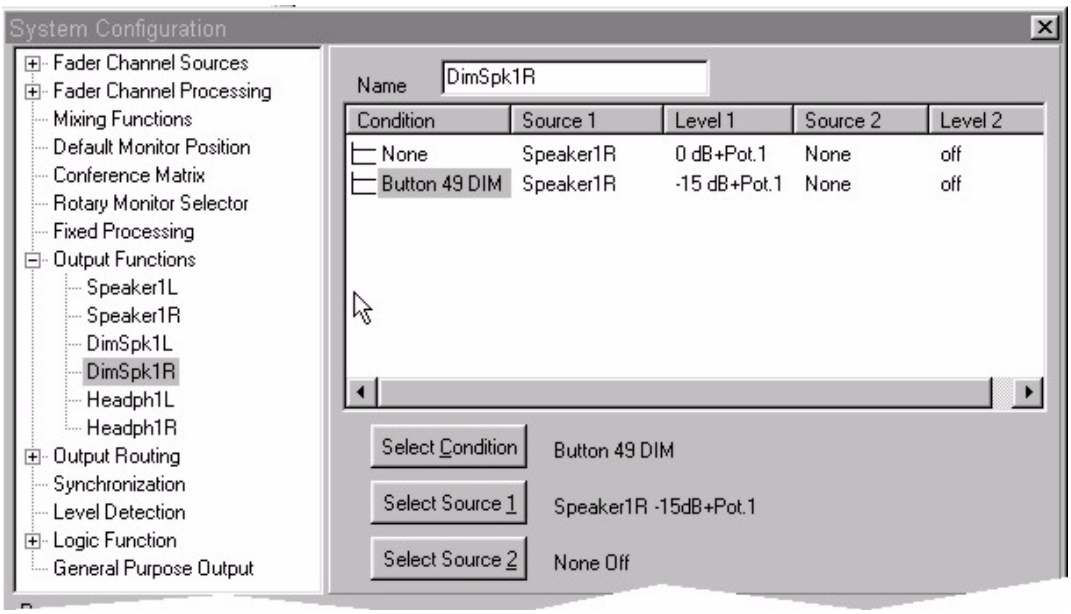

Figure 62: Condition list “DimSpk1R”

Output Function 5 “Headph1L”

Condition list	Source 1	Level 1	Source 2	Level 2
None	Monitor 1 L	0 dB + Pot.2	PFL L	off
PFL on	Monitor 1 L	off	PFL L	0 dB + Pot.2

Table 9: Condition list “Headph1L”

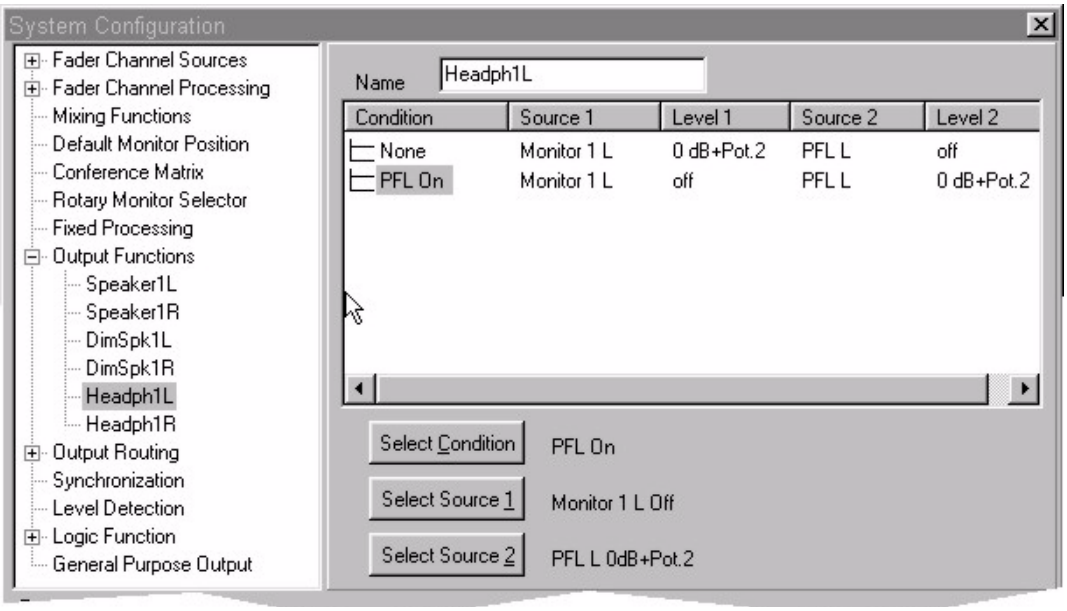

Figure 63: Condition list “Headph1L”

Output Function 6 “Headph1R”

Condition list	Source 1	Level 1	Source 2	Level 2
None	Monitor 1 R	0 dB + Pot.2	PFL R	off
DIM + PFL on	Monitor 1 R	off	PFL R	0 dB + Pot.2

Table 10: Condition list “Headph1R”

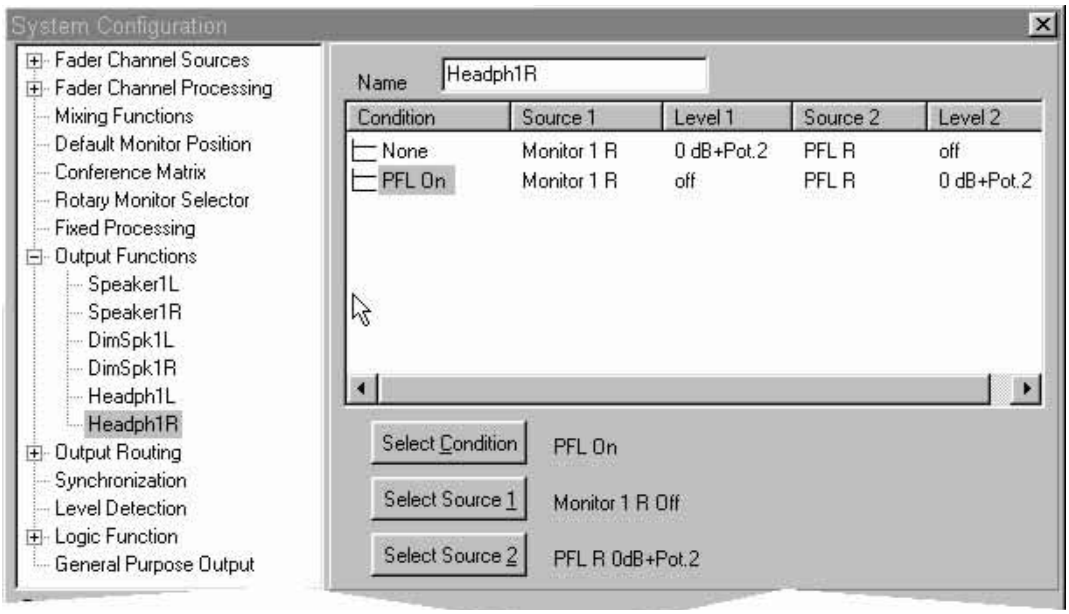

Figure 64: Condition list “Headph1R”

3.2.3 Monitor Busses

The ***Monitor Busses*** are special routing functions (no special DSP resources!) to build monitor selectors on the control desk. Six Stereo Monitor Busses are available on the RM3200D.

The application is to build ***Interlocking Monitor Selectors*** with push buttons or with the rotary control on the Main Module. More than one Monitor Buss is useful in larger systems, for example:

- Monitor Bus 1 for the Control Room Loudspeakers
- Monitor Bus 2 for an independent Metering
- Monitor Bus 3 for Studiomonitoring
- etc.

For the PFL function we do not use one of the Monitor Busses 1 to 6, because PFL is commonly used as summing PFL, not as interlocked. Refer to the **PFL** section.

Important Note: The PFL Bus is *not* a Monitor Bus, the PFL Bus is a special Summing Bus!

In **Figure 58 on page 64** you find an example for a simple Monitor Bus. To select the four monitor sources Tuner, DAT, Pgm 1 and Pgm 2 you need:

1. Four push buttons on the Main Module or Studio Unit of the control desk to build the interlocking selector push buttons
2. and one Monitor Bus.

Definition.exe

In the configuration software you must assign the `Function: Monitor Channel` to the four push buttons and select the number of the Monitor Bus, in this example `Monitornumber: 1`.

After that, assign the source for the left and the right channel of the Monitor Bus by pressing the buttons `Select Left` and select the source in the window `Audio Source` from the TDM Bus.

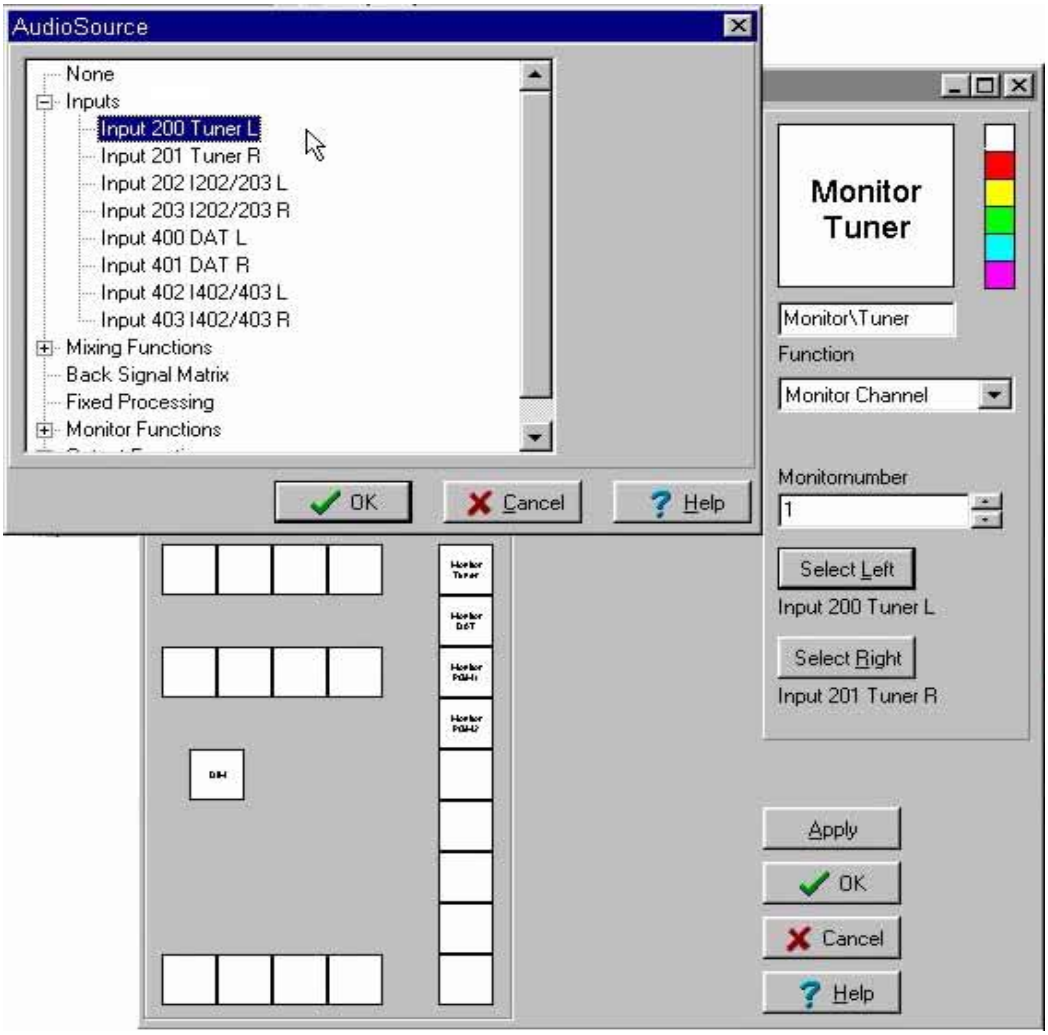

Figure 65: Monitor Selector

Do this for all four push buttons and see the **example "PFL Monitor" on the pages before**, how to use the Monitor Bus together with Output Functions to build the monitoring of the mixing console, normally after the Monitor Bus is an Output Function to control the levels and to create other functions like PFL/Monitor split or talkback functions.

Applications to use the Monitor Bus directly routed to an output are for example the connection of meters (PPMs) or the connection of your own headphone amplifiers with an integrated volume control.

3.3 Synchronization

The synchronization of the RM3200D is very powerful. Several sync sources are available:

- Internal sync 48kHz on DSP Backplane 1 - this is always the default sync source for the system
- Internal sync 44.1kHz on DSP Backplane 1.

Because the internal sync sources on the DSP Backplanes do not match the requirements of the AES recommendations, a higher accuracy clock (±10ppm) is available on slot 1 with the MADI Module RM330-421 or with the Sync Module RM330-410.

Also, on these two modules several external sync sources are available:

- TTL Wordclock on the BNC connector
- AES3/EBU on the BNC connector (only on the RM330-410)
- MADI (only on the RM330-421).

For each of the external sync sources, three PLL circuits are available to match your requirements:

- A free running RC-PLL circuit with a locking range of 30kHz to 48,5kHz
- A high accuracy Quartz-PLL circuit for 48kHz (±75ppm locking range) with a high jitter rejection
- A high accuracy Quartz-PLL circuit for 44,1kHz (+-75ppm locking range) with a high jitter rejection

At least, the Digital Input Modules RM330-110 and the slots 2 to 9 can supply the system with a sync coming from a connected AES3/EBU signal. This function is not available with the Module RM330-111.

Important Note: The synchronization functions for the RM3200D DSP Frames are only available on the first DSP Backplane or on the first 10 slots. The DSP Backplanes 2 and 3 (slots 11 to 30) are always slaves to the DSP Backplane 1.

The different sync sources can be configured as Sync Source 1 and Sync Source 2 in the configuration software. When the Sync Source 1 is not longer available, the system switches automatically to the Sync Source 2. When the Sync Sources 1 and 2 both are not available, the system switches automatically to the default sync Internal 48kHz.

There is one restriction: When you work with an external sync source, the sync must be valid when switching on the RM3200D or when restarting the system. Otherwise, the system switches to the internal 48kHz and will not fall back when Sync Source 1 becomes valid!

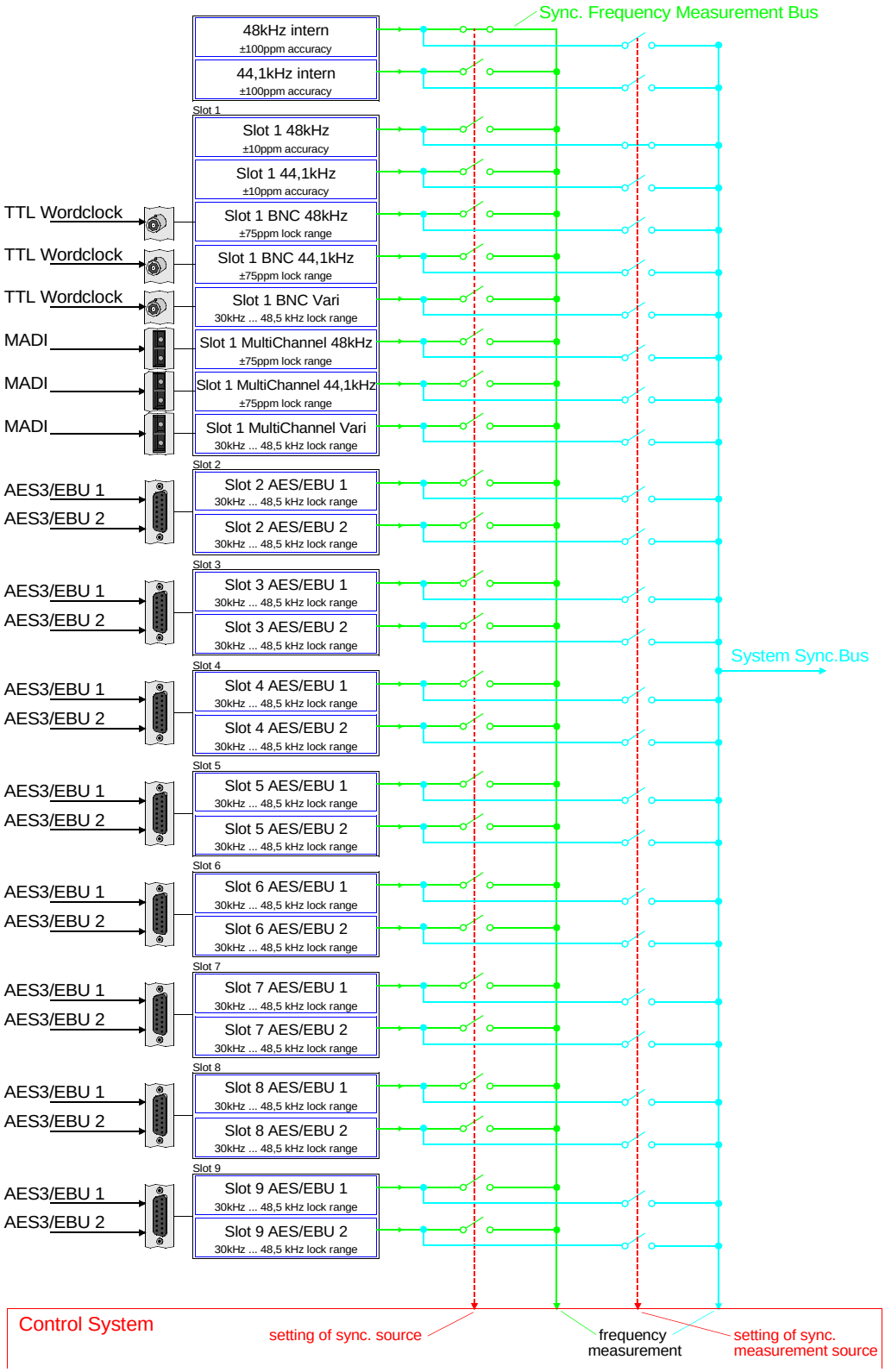

Figure 66: slot_madi

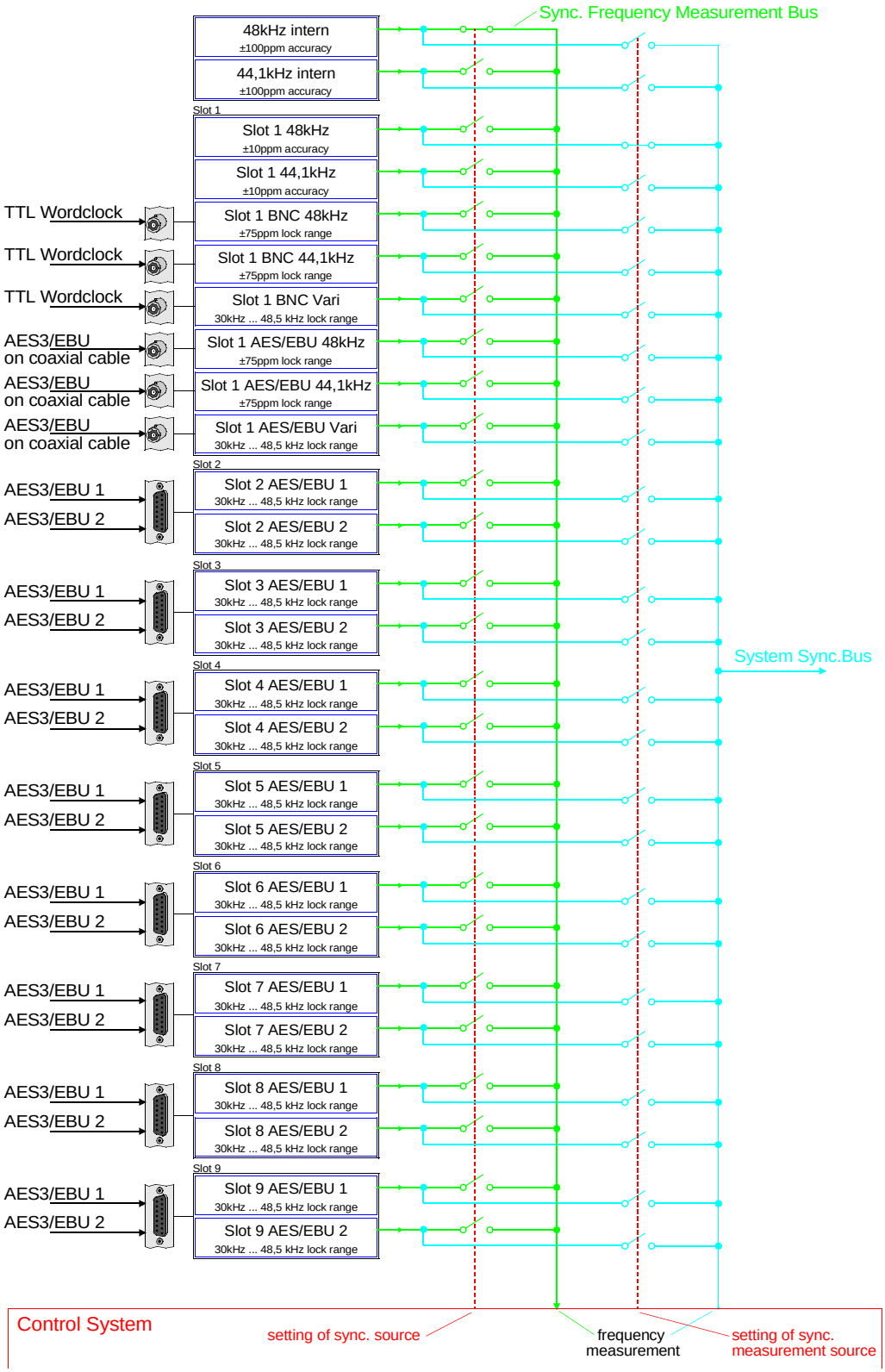

Figure 67: slot_sync

3.4 Logic functions

A maximum of 50 logic functions is available with max. 20 input signals each. Each logic function can work either as “AND” or as “OR”.

Inserting a logic function

Definition.exe

1. Select the tree node “`Logic Function`”.
2. Select the new logic function from the popup menu “Insert new Logicfunction” either by pressing the right mouse button or the Insert-key.

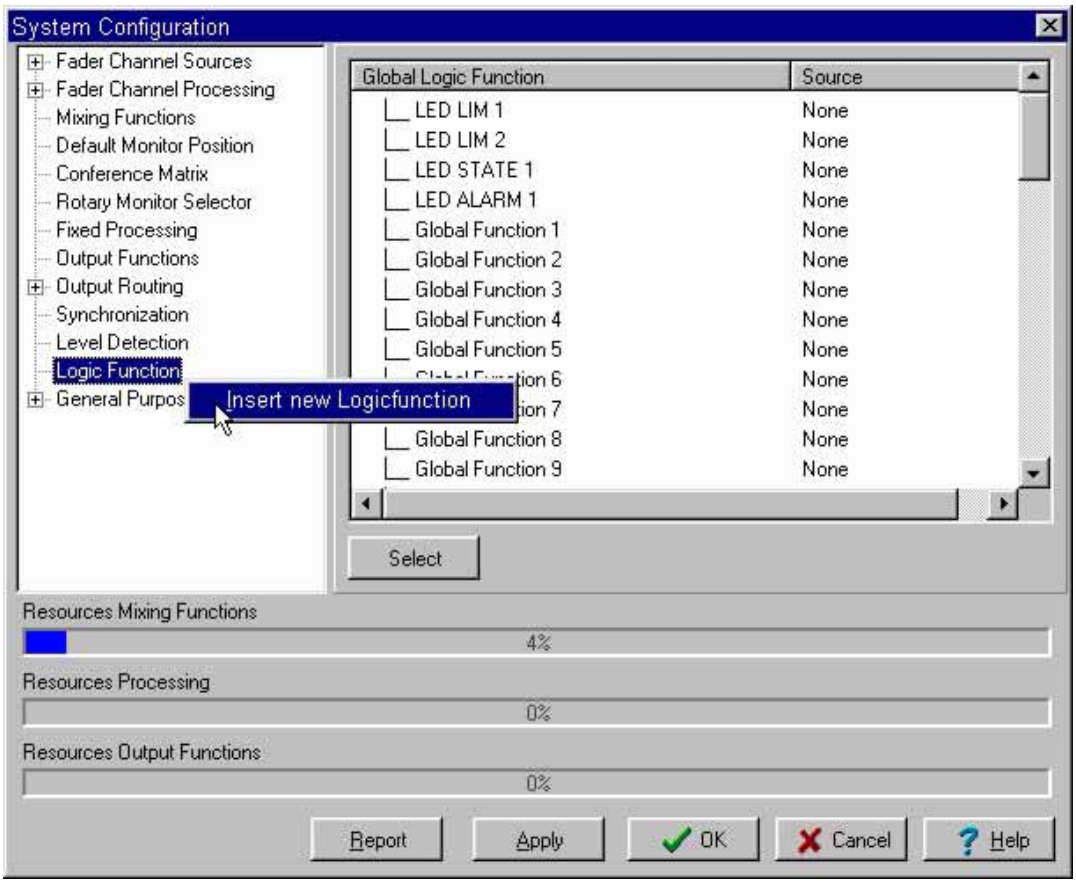

Figure 68:

Assigning logical operations

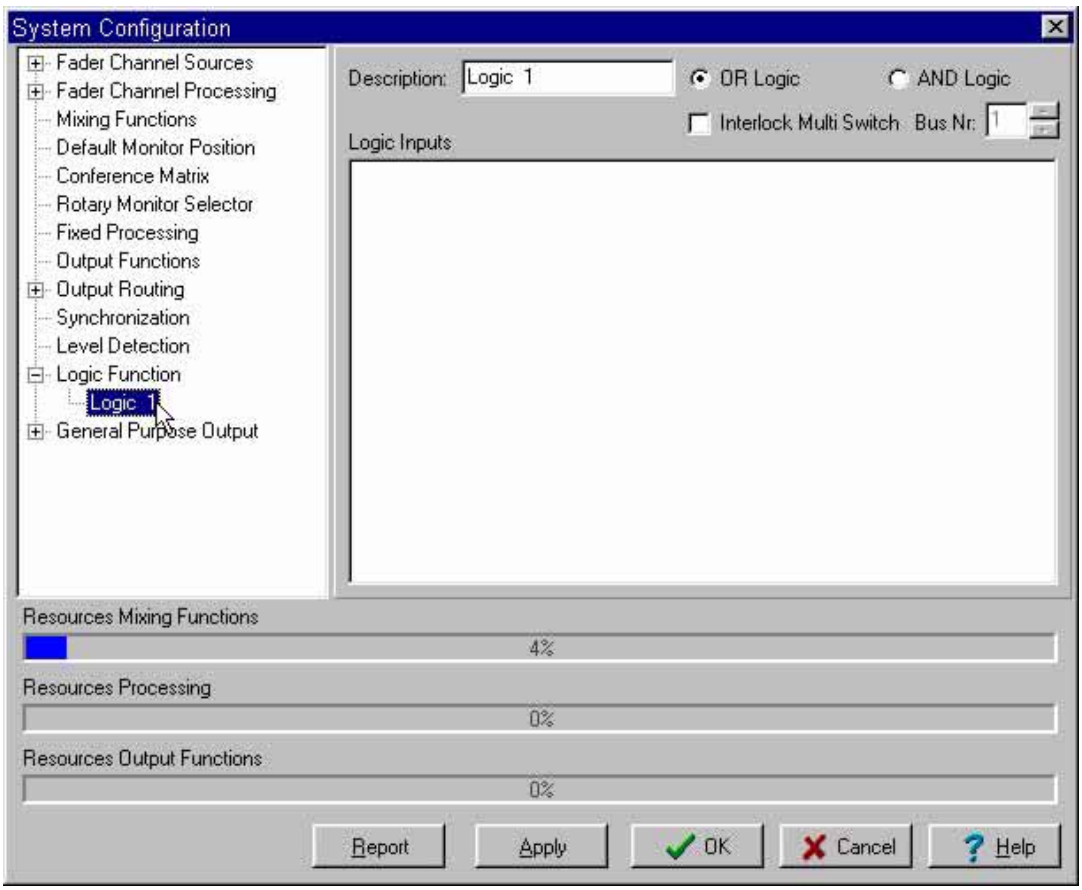

Figure 69:

1. The name of the logic function can be entered into the box “Description” (10 characters max.).
2. The boxes “OR Logic” and “AND Logic” determine the type of the operation.

Clicking the right mouse button in the box “Logic Inputs” or pressing the Insert-key on your keyboard to adds a new input signal.

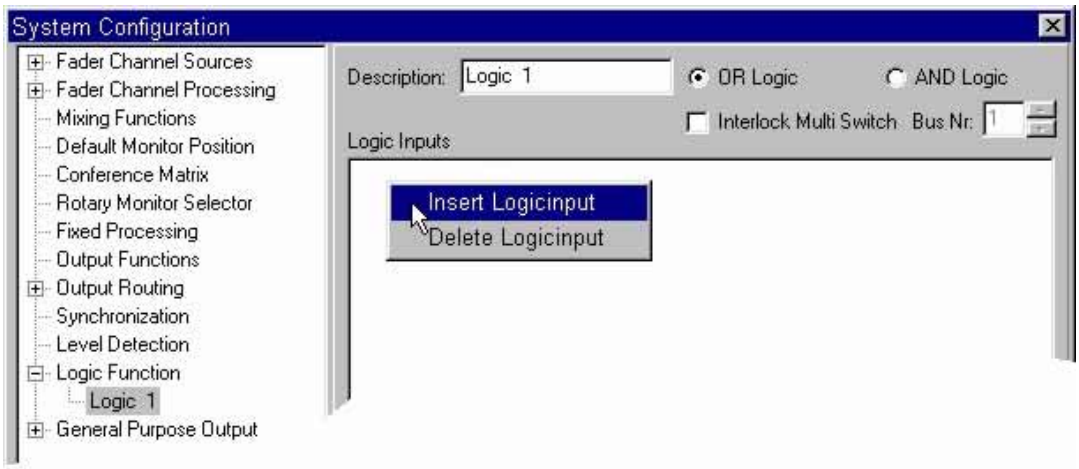

Figure 70:

The following input signals are available:

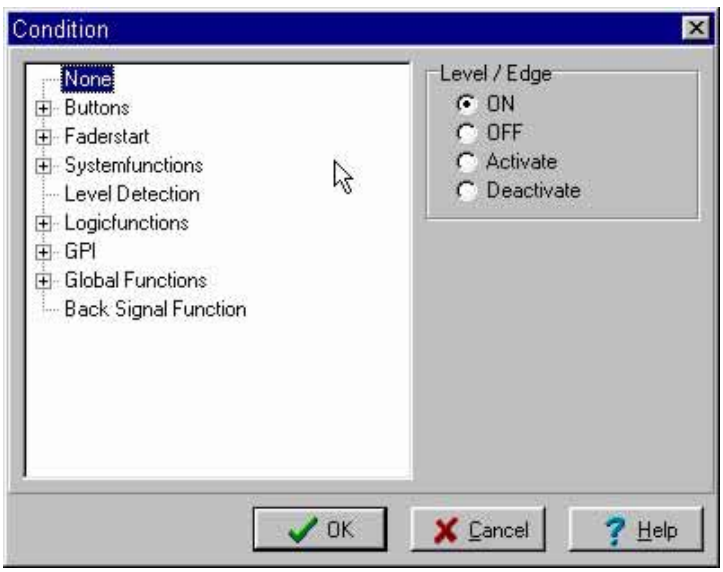

Figure 71:

- All keys of the main control panel (independent from other functions assigned to a key)
- All fader start contacts of the faders (independent from fader assignment to channels)
- All remote starts of the inputs (dependent on the set fader start level and the fader)
- System functions (3 pulse generators with different frequencies)
- All outputs of the logic functions
- All optical switch inputs

For each signal, the following can be defined:

- Static on (ON)

- Static off (OFF)
- Pulse when activating the signal
- Pulse when deactivating the signal

Example 1

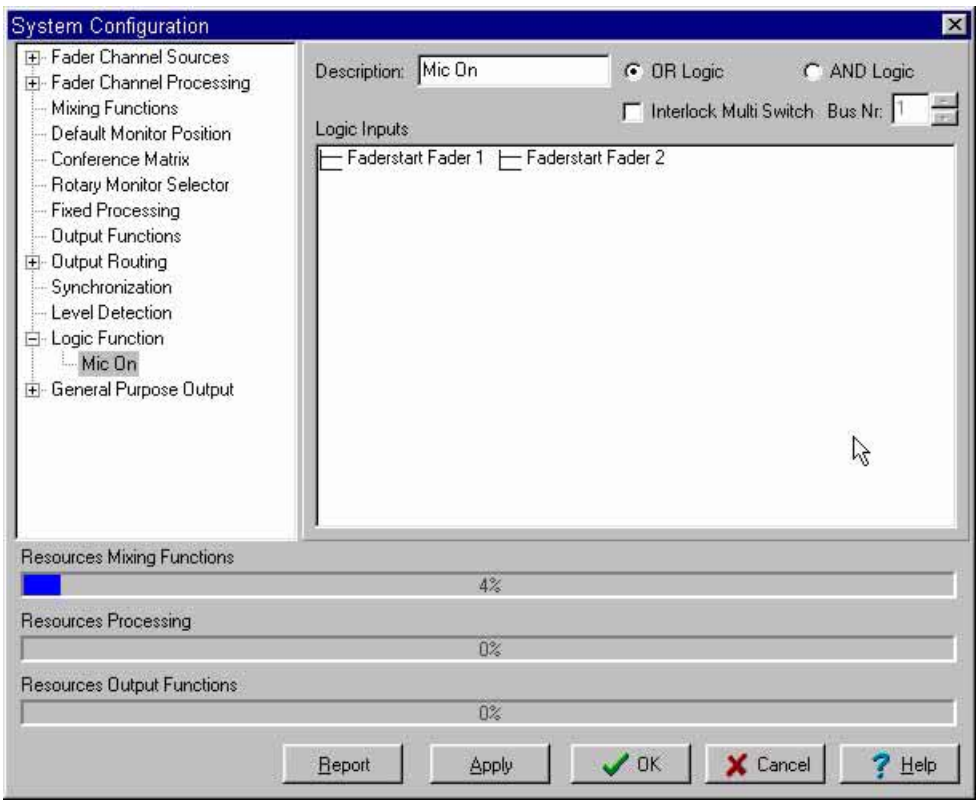

Figure 72:

In this example, the logic function “On-Air-Light” was configured. This function becomes active, when “Faderstart Fader 1” or “Faderstart Fader 2” is active.

Example 2

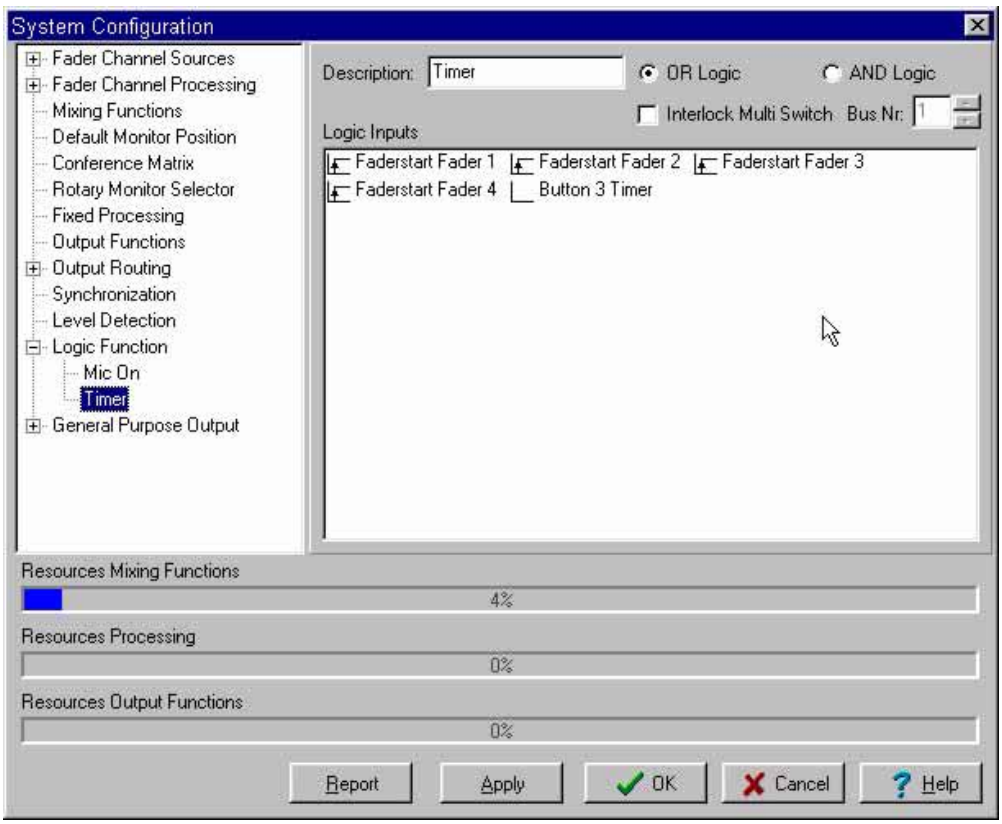

Figure 73:

In this example, the logic function “Timer” was configured. This function is to reset a clock when opening a fader. If key 3 “Timer” is not active, the clock is stopped permanently.