Partner: L-Acoustics
Model: P1
Device Type: Networked Audio Processor

GENERAL INFORMATION

SIMPL WINDOWS NAME:	L-Acoustics Networked Audio Processor
CATEGORY:	AUDIO PROCESSOR
VERSION:	V3.2.0
SUMMARY:	The module provides control and configuration functions via TCP/IP.
GENERAL NOTES:	This module is for the control of the P1 networked AVB audio processors. Each Processor to be connected and controlled requires one module instance assigned to it. Each module should therefore use unique digital, analog and serial joins. The simplest way of achieving this is to use a unique prefix which identifies the processor such as PROC1_MUTE and PROC2_MUTE.
CRESTRON HARDWARE REQUIRED:	C3ENET, 3-Series Processor
SETUP OF CRESTRON HARDWARE:	The Crestron processor's IP address must be in the same subnet as the L-Acoustics processors (typically 192.168.1.x/255.255.255.0, but other classes are possible, see networked audio processor user manual, or IpAddress parameter description). If not the case, then the TCP/IP connection will be impossible, as L-Acoustics networked audio processors currently don't support Layer 3 IP routing.
VENDOR FIRMWARE:	P1 minimum firmware version: 2.9.1.x Maximum firmware version: 2.10.x
VENDOR SETUP:	Networked Audio Processor connected to the Ethernet Network

SUPPORT CONTACT

COMPANY NAME:	L-Acoustics
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RELEASE NOTES

- **Version 3.2.0** *(version number aligned with Amplified Controller modules)*

New features/Improvements

0007733	Support of Firmware 2.11.x

Fixed issues

0007707	Digital inputs don't work well with 0 and 1 standard signals

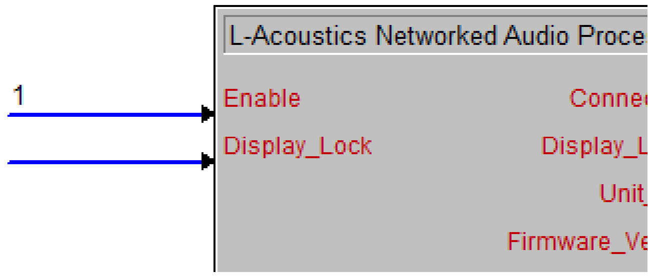
- **Version 1.1.0 (April 2020)**
  - Support of Firmware 2.10.x
  - Support P1 Busses 5 to 8
  - Control of P1 USB Media Player

- **Version 1.0.0** (February 2019)

Initial release

GENERAL INFORMATION

- **Standard signals "1" and "0"**



It is possible to use the standard "**1**" signal on all digital inputs of the module, especially on the **Enable** input. This will have the effect to instantly connect the module at program startup (for **Enable** input), or send the associated command as soon as the module connection is established (for other digital inputs).

This is also true for analog initialized signals.

However, the stand signal "**0**" cannot be used to automatically trigger the falling edge of a command. Using **"0"** on an input has the same effect as commenting **"//"** the input.

CONTROL

Input Signal Name	Type	Description
Enable	D	The **Enable** signal is used to activate the functions of the module. As soon as this signal is HIGH, the module tries to connect to the networked audio processor over TCP/IP. When the connection is successful, all other input signals are effective. When the signal is LOW, the TCP/IP connection gets closed, and input signals become ineffective.
Display_Lock	D	**Display_Lock** and **Set_Standby** react to rising edges. Setting these signals HIGH turns the networked audio processor into either Online state or Standby state. Of course, it cannot be in both states at the same time, so the latest rising edge will prevail.
Mute_ANA_in*[X]* **Mute_AES_in*[X]*** **Mute_AVB_in*[X]*** **Mute_MIC_in*[X]***	D	**Mute_*[YYY]*_in*[X]*** control the mute state of the associated input channels. *[X]* ranges between 1 and 4 or 1 and 8 depending on the input type. HIGH = input channel is muted LOW = input channel is unmuted
Mute_BUS_*[X]*	D	**Mute_BUS_*[X]*** control the mute state of the associated DSP busses. *[X]* ranges between 1 and 8. HIGH = DSP bus is muted LOW = DSP bus is unmuted
Mute_ANA_out*[X]* **Mute_AES_out*[X]*** **Mute_AVB_out*[X]***	D	**Mute_*[YYY]*_out*[X]*** control the mute state of the associated output channels. *[X]* ranges between 1 and 4 or 1 and 8 depending on the output type. HIGH = output channel is muted LOW = output channel is unmuted
Mute_MPL **Mute_GEN**	D	**Mute_*[YYY]*** control the mute state of the associated internal audio generators. GEN is the internal signal/noise generator, MPL is the USB Media Player. HIGH = audio is muted LOW = audio is unmuted
Gain_ANA_in*[X]*# **Gain_AES_in*[X]*#** **Gain_AVB_in*[X]*#** **Gain_MIC_in*[X]*#**	A	**Gain_*[YYY]*_in*[X]*#** control the gain value of the associated input channels. *[X]* ranges between 1 and 4 or 1 and 8 depending on the input type. The full analog signal range (0d ~ 65535d) is used to represent the full gain range (−60.0 ~ +15.0dB) of the input channels, which means that: - **Gain_*[YYY]*_in*[X]*#** = 0d (minimum) ⇒ gain = −60.0dB (minimum) - **Gain_*[YYY]*_in*[X]*#** = 65535d (maximum) ⇒ gain = +15.0dB (maximum) - **Gain_*[YYY]*_in*[X]*#** = 52428d ⇒ gain = 0.0dB (unity) The analog signal value is immediately applied to the associated input channel.

Input Signal Name	Type	Description
Gain_BUS_*[X]*#	A	**Gain_BUS_*[X]*#** control the gain value of the associated DSP busses. *[X]* ranges between 1 and 8. The full analog signal range (0d ~ 65535d) is used to represent the full gain range (−60.0 ~ +15.0dB) of the DSP busses, which means that: - **Gain_BUS_*[X]*#** = 0d (minimum) ⇒ gain = −60.0dB (minimum) - **Gain_BUS_*[X]*#** = 65535d (maximum) ⇒ gain = +15.0dB (maximum) - **Gain_BUS_*[X]*#** = 52428d ⇒ gain = 0.0dB (unity) The analog signal value is immediately applied to the associated DSP bus.
Gain_ANA_out*[X]*# **Gain_AES_out*[X]*#** **Gain_AVB_out*[X]*#**	A	**Gain_*[YYY]*_out*[X]*#** control the gain value of the associated output channels. *[X]* ranges between 1 and 4 or 1 and 8 depending on the input type. The full analog signal range (0d ~ 65535d) is used to represent the full gain range (−60.0 ~ +15.0dB) of the output channels, which means that: - **Gain_*[YYY]*_out*[X]*#** = 0d (minimum) ⇒ gain = −60.0dB (minimum) - **Gain_*[YYY]*_out*[X]*#** = 65535d (maximum) ⇒ gain = +15.0dB (maximum) - **Gain_*[YYY]*_out*[X]*#** = 52428d ⇒ gain = 0.0dB (unity) The analog signal value is immediately applied to the associated output channel.
Gain_MPL# **Gain_GEN#**	A	**Gain_*[YYY]*#** control the gain value of the associated internal audio generators. GEN is the internal signal/noise generator, MPL is the USB Media Player. The full analog signal range (0d ~ 65535d) is used to represent the full gain range (−60.0 ~ +0.0dB) of the internal generators, which means that: - **Gain_*[YYY]*#** = 0d (minimum) ⇒ gain = −60.0dB (minimum) - **Gain_*[YYY]*#** = 65535d (maximum) ⇒ gain = +0.0dB (maximum) The analog signal value is immediately applied to the associated internal audio generator.
Preamp_Gain_MIC_in*[X]*#		**Preamp_Gain_MIC_in*[X]*#** control the preamp gain value of the front MIC/LINE input channels. *[X]* range from 1 to 4. Preamp gain possible values are by 3dB steps, from +0dB to +60dB. Analog signal possible values range from 0d to 60d, the final choice being rounded to the closest valid preamp gain (multiple of 3). e.g. - **Preamp_Gain_MIC_in*[X]*#** = 0d ⇒ gain = +0dB (minimum) - **Preamp_Gain_MIC_in*[X]*#** = 5d ⇒ gain = +6dB (rounded) - **Preamp_Gain_MIC_in*[X]*#** = 6d ⇒ gain = +6dB - **Preamp_Gain_MIC_in*[X]*#** = 7d ⇒ gain = +6dB (rounded) - **Preamp_Gain_MIC_in*[X]*#** = 80d ⇒ gain = +60dB (maximum) The analog signal value is immediately applied to the associated MIC/LINE input channel.

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Input Signal Name	Type	Description
AES12_Fallback_Source#	A	**AES12_Fallback_Source#** control the enablement and source selection for automatic failover of inputs AES 1/2 to analog inputs. List of possible values: - 0d : fallback is disabled for AES 1/2 - 1d : fallback is enabled, and using ANA 1/2 as failover sources - 2d : fallback is enabled, and using MIC 1/2 as failover sources Setting this input to 0d instantly resets the fallback state, and restores AES 1/2 as the effective input sources.
AES34_Fallback_Source#	A	**AES34_Fallback_Source#** control the enablement and source selection for automatic failover of inputs AES 3/4 to analog inputs. List of possible values: - 0d : fallback is disabled for AES 3/4 - 1d : fallback is enabled, and using ANA 3/4 as failover sources - 2d : fallback is enabled, and using MIC 3/4 as failover sources Setting this input to 0d instantly resets the fallback state, and restores AES 3/4 as the effective input sources.
AVB14_Fallback_Source# **AVB58_Fallback_Source#**	A	**AVB*[XX]*_Fallback_Source#** control the enablement and source selection for automatic failover of inputs AVB 1 ~ 4 and AVB 5 ~ 8 to XLR inputs. List of possible values: - 0d : fallback is disabled for AVB*[XX]* - 1d : fallback is enabled, and using ANA 1 ~ 4 as failover sources - 2d : fallback is enabled, and using MIC 1 ~ 4 as failover sources - 3d : fallback is enabled, and using AES 1 ~ 4 as failover sources Setting this input to 0d instantly resets the fallback state, and restores AVB 1 ~ 4 and AVB 5 ~ 8 as the effective input sources.
AES12_Fallback_Reset **AES34_Fallback_Reset** **AVB14_Fallback_Reset** **AVB58_Fallback_Reset**	D	**[YYY][XX]_Fallback_Reset** react to rising edges. Push these signals to restore the main audio sources in case when fallback was activated (main source in error) and now the main source is valid again. Pushing **AES12_Fallback_Reset** (resp. **AES34_Fallback_Reset**) has no effect if input AES 1/2 (resp. AES 3/4) is still in error, as fallback is immediately reactivated. ⚠ However, pushing **AVB14_Fallback_Reset** or **AVB58_Fallback_Reset** is always deactivating fallback (resetting AVB 1 ~ 4 or AVB 5 ~ 8 as effective source), even if AVB input stream is not valid.
AES12_Fallback_Trigger **AES34_Fallback_Trigger** **AVB_Fallback_Trigger**	D	**[YYY][XX]_Fallback_Trigger** react to rising edges. Push these signals to manually trigger fallback. Pushing these signals have no effect if fallback is not enabled, or is enabled and already active.

Input Signal Name	Type	Description
Configuration_Load#	A	**Configuration_Load#** accepts values between 1d and 30d. When changing this value, if a configuration is available in the corresponding slot of the networked audio processor, then this configuration is loaded.
Source_ANA_out*[X]*# **Source_AES_out*[X]*#** **Source_AVB_out*[X]*#** **Source_MON_*[X]*#**	A	**Source_*[YYY]*_out*[X]*#** and **Source_MON_*[X]*#** control the routing of internal and external audio sources to output channels and headphones (MON), and accept values between 0d and 27d. List of possible values for each output channel or headphones (MON): - 0d : NONE (silent) - 1d ~ 4d : ANALOG input channels 1 ~ 4 - 5d ~ 8d : AES/EBU input channels 1 ~ 4 - 9d ~ 16d : AVB input channels 1 ~ 8 - 17d ~ 20d : MIC/LINE input channels 1 ~ 4 - 21d ~ 24d : DSP busses 1 ~ 4 - 25d : CUE bus - 26d : internal signal/noise generator - 27d ~ 28d : internal Media Player L ~ R - 29d ~ 32d : DSP busses 5 ~ 8 Setting the input signal to an unknown value selects the default 0d (silent).
Generator_Enable	D	HIGH = enable the internal signal/noise generator LOW = disable the internal signal/noise generator Note: enabling the internal signal/noise generator forces its signal type to Sine.
Generator_Frequency#	A	**Generator_Frequency#** accepts values between 1d and 24000d. This signal is setting the internal generator's frequency parameter of 'Sine' type. The value is directly converted to Hz (0 Hz ~ 24000 Hz).
Generator_Mix_BUS_*[X]*	D	**Generator_Mix_BUS_*[X]*** control the mixing of internal signal/noise generator into the DSP busses 1 ~ 8. HIGH = the generator signal is mixed to BUS *[X]* at 0dB level LOW = the generator signal is not mixed to BUS *[X]*
GPO_*[X]*	D	**GPO_[X]** control the opening/closing of the processor's GPO relays. HIGH = GPO [X] relays is closed LOW = GPO [X] relay is open
Player_Play	D	**Player_Play** controls the play/pause state of the processor's USB Media Player HIGH = Playing LOW = Paused
Player_Previous_Track **Player_Next_Track**	D	**Player_Previous_Track** and **Player_Next_Track** react to rising edges. They instantly load the previous (resp. next) track in the Media Player folder, if any.

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Input Signal Name	Type	Description
Player_Load_Track#	A	**Player_Load_Track#** accepts values between 1d and **Player_Tracks_Count#** (see FEEDBACK section). When changing this value, if a track is available in the Media Player folder at this index, then it is loaded. Playback is paused after loading the track, ready to play from the beginning of the track.
Player_Playback_Mode	D	**Player_Playback_Mode** configures how the Media Player is bahving when a track is finished. HIGH = File mode: play a single track LOW = Folder mode: play all tracks in the folder
Player_Repeat_Mode	D	**Player_Repeat_Mode** configures how the Media Player is repeating tracks (either a single track, or the whole folder, see **Player_Playback_Mode**). HIGH = Repeat enabled: track of folder played in loop LOW = Repeat disabled: track or folder played once

FEEDBACK

Output Signal Name	Type	Description
Connected_fb	D	This signal is HIGH when the TCP/IP connection to the networked audio processor is established, and the remote device is compatible with the module.
Display_Lock_fb	D	HIGH = the front panel controls are locked. LOW = the front panel controls are unlocked.
Unit_Type$	S	This signal represents the amplified controller type connected by the module. Possible value is only 'P1' for the moment.
Unit_Ip_Address$	S	IP address of the connected unit (primary IP address if unit is in redundant network mode). Example: '192.168.1.100'
Firmware_Version$	S	This signal represents the networked audio processor's current version of firmware. Example: '2.9.3.4'
Mute_ANA_in*[X]*_fb **Mute_AES_in*[X]*_fb** **Mute_AVB_in*[X]*_fb** **Mute_MIC_in*[X]*_fb**	D	**Mute_*[YYY]*_in*[X]*** represent the current mute state of the associated input channels. *[X]* ranges between 1 and 4 or 1 and 8 depending on the input type. HIGH = input channel is muted LOW = input channel is unmuted
Mute_BUS_*[X]*_fb	D	**Mute_BUS_*[X]*** represent the current mute state of the associated DSP busses. *[X]* ranges between 1 and 8. HIGH = DSP bus is muted LOW = DSP bus is unmuted
Mute_ANA_out*[X]*_fb **Mute_AES_out*[X]*_fb** **Mute_AVB_out*[X]*_fb**	D	**Mute_*[YYY]*_out*[X]*** represent the current mute state of the associated output channels. *[X]* ranges between 1 and 4 or 1 and 8 depending on the output type. HIGH = output channel is muted - LOW = output channel is unmuted
Mute_MPL_fb **Mute_GEN_fb**	D	**Mute_*[YYY]*** represent the current mute state of the associated internal audio generators. GEN is the internal signal/noise generator, MPL is the USB Media Player. HIGH = audio is muted LOW = audio is unmuted

Output Signal Name	Type	Description
Gain_ANA_in*[X]*_fb# **Gain_AES_in*[X]*_fb#** **Gain_AVB_in*[X]*_fb#** **Gain_MIC_in*[X]*_fb#**	A	***Gain_[YYY]_in[X]#*** represent the current gain value of the associated input channels. *[X]* ranges between 1 and 4 or 1 and 8 depending on the input type. The full analog signal range (0d ~ 65535d) is used to represent the full gain range (−60.0 ~ +15.0dB) of the input channels, which means that: - ***Gain_[YYY]_in[X]#*** = 0d (minimum) ⇒ gain = −60.0dB (minimum) - ***Gain_[YYY]_in[X]#*** = 65535d (maximum) ⇒ gain = +15.0dB (maximum) - ***Gain_[YYY]_in[X]#*** = 52428d ⇒ gain = 0.0dB (unity)
Gain_BUS_*[X]*_fb#	A	**Gain_BUS_*[X]*#** represent the current gain value of the associated DSP busses. *[X]* ranges between 1 and 8. The full analog signal range (0d ~ 65535d) is used to represent the full gain range (−60.0 ~ +15.0dB) of the DSP busses, which means that: - **Gain_BUS_*[X]*#** = 0d (minimum) ⇒ gain = −60.0dB (minimum) - **Gain_BUS_*[X]*#** = 65535d (maximum) ⇒ gain = +15.0dB (maximum) - **Gain_BUS_*[X]*#** = 52428d ⇒ gain = 0.0dB (unity)
Gain_ANA_out*[X]*_fb# **Gain_AES_out*[X]*_fb#** **Gain_AVB_out*[X]*_fb#**	A	***Gain_[YYY]_out[X]#*** represent the current gain value of the associated output channels. *[X]* ranges between 1 and 4 or 1 and 8 depending on the input type. The full analog signal range (0d ~ 65535d) is used to represent the full gain range (−60.0 ~ +15.0dB) of the output channels, which means that: - ***Gain_[YYY]_out[X]#*** = 0d (minimum) ⇒ gain = −60.0dB (minimum) - ***Gain_[YYY]_out[X]#*** = 65535d (maximum) ⇒ gain = +15.0dB (maximum) - ***Gain_[YYY]_out[X]#*** = 52428d ⇒ gain = 0.0dB (unity)
Gain_MPL_fb# **Gain_GEN_fb#**	A	***Gain_[YYY]#*** represent the current gain value of the associated internal audio generators. GEN is the internal signal/noise generator, MPL is the USB Media Player. The full analog signal range (0d ~ 65535d) is used to represent the full gain range (−60.0 ~ +0.0dB) of the internal generators, which means that: - ***Gain_[YYY]#*** = 0d (minimum) ⇒ gain = −60.0dB (minimum) - ***Gain_[YYY]#*** = 65535d (maximum) ⇒ gain = +0.0dB (maximum)
Preamp_Gain_MIC_in*[X]*_fb#	A	**Preamp_Gain_MIC_in*[X]*#** represents the current preamp gain value of the front MIC/LINE input channels. *[X]* range from 1 to 4. Preamp gain possible values are by 3dB steps, from +0dB to +60Db (analog = 0d ~ 60d).

Output Signal Name	Type	Description
AES12_Fallback_Source_fb#	A	**AES12_Fallback_Source#** represents the current enablement and source selection for automatic failover of inputs AES 1/2 to analog inputs. List of possible values: - 0d : fallback is disabled for AES 1/2 - 1d : fallback is enabled, and using ANA 1/2 as failover sources - 2d : fallback is enabled, and using MIC 1/2 as failover sources
AES34_Fallback_Source_fb#	A	**AES34_Fallback_Source#** represents the current enablement and source selection for automatic failover of inputs AES 3/4 to analog inputs. List of possible values: - 0d : fallback is disabled for AES 3/4 - 1d : fallback is enabled, and using ANA 3/4 as failover sources - 2d : fallback is enabled, and using MIC 3/4 as failover sources
AVB14_Fallback_Source_fb# **AVB58_Fallback_Source_fb#**	A	**AVB*[XX]*_Fallback_Source#** represents the current enablement and source selection for automatic failover of inputs AVB 1 ~ 4 and AVB 5 ~ 8 to XLR inputs. List of possible values: - 0d : fallback is disabled for AVB*[XX]* - 1d : fallback is enabled, and using ANA 1 ~ 4 as failover sources - 2d : fallback is enabled, and using MIC 1 ~ 4 as failover sources - 3d : fallback is enabled, and using AES 1 ~ 4 as failover sources
AES12_Fallback_Active_fb **AES34_Fallback_Active_fb** **AVB14_Fallback_Active_fb** **AVB58_Fallback_Active_fb**	D	***[YYY][XX]*_Fallback_Active_fb** represents the current fallback state of input channels. HIGH = fallback is active: the backup input channels are used as audio source LOW = fallback is inactive: the main input channels are used as audio source

Output Signal Name	Type	Description
Meters_ANA_in# **Meters_AES_in#** **Meters_MIC_in#** **Meters_AVB_in#** **Meters_MPL#** **Meters_BUS#** **Meters_ANA_out#** **Meters_AES_out#** **Meters_AVB_out#**	A	These analog signals are bit-fields representing the presence of audio signal within the associated channel, bus or generator. For ANALOG, AES/EBU, AVB and MIC/LINE: - bit0 = channel 1 - bit1 = channel 2 - bit2 = channel 3 - bit3 = channel 4 For AVB: - bit4 = channel 5 - bit5 = channel 6 - bit6 = channel 7 - bit7 = channel 8 For DSP busses: - bit0 = Bus 1 - bit1 = Bus 2 - bit2 = Bus 3 - bit3 = Bus 4 - bit4 = Bus 5 - bit5 = Bus 6 - bit6 = Bus 7 - bit7 = Bus 8 For Media Player: - bit0 = Stereo left channel - bit1 = Stereo right channel Bit HIGH = the channel/bus/generator has audio level >−60.0dBFS Bit LOW = the channel/bus/generator has audio level ≤ −60.0dBFS
Meters_GEN	D	Bit HIGH = the generator has audio level >−60.0dBFS Bit LOW = the generator has audio level ≤ −60.0dBFS
Configuration_Load_fb#	A	**Configuration_Load_fb#** reflects the slot number of the currently loaded configuration on the processor. The value is between 1d and 30d.
Current_Configuration_Name$	S	**Current_Configuration_Name$** represents the name of the currently loaded configuration on the processor. The name includes the slot number, and is prefixed with an asterisk (*) if some parameters have been changed since last load of configuration. Examples: - 01:DEFAULT - *05:FOH
Configuration_Names_List$	S	**Configuration_Names_List$** contains a concatenated list of all configuration names available in the processor. Names contain the prefixed slot number, and are separated by a carriage return character (`\r`, or `\x0D`). e.g.: "`01:CONFIG1\r02:CONFIG2\r03CONFIG3\r`…"

Output Signal Name	Type	Description
AES12_Error **AES34_Error** **AVB_Error**	D	HIGH = input unlocked or in error state LOW = input is locked and valid
AES12_Message$** **AES34_Message$ **AVB_Message$**	S	These signals accompany the **AES12_Error**, **AES34_Error** and **AVB_Error** signals, and give a text description of what is failing with the input status.
Error_Present	D	HIGH = errors (other than input errors) are currently raised LOW = all errors are cleared
Error_Messages$	S	This signal accompanies the **Error_Present** signal, and gives a text description of what is failing in the networked audio processor.
Source_ANA_out*[X]*_fb# **Source_AES_out*[X]*_fb#** **Source_AVB_out*[X]*_fb#** **Source_MON_*[X]*_fb#**	A	**Source_*[YYY]*_out*[X]*_fb#** and **Source_MON_*[X]*_fb#** represent the current routing of internal and external audio sources to output channels and headphones (MON). Check out the description of **Source_*[YYY]*_out*[X]*#** for the possible values for these signals.
Generator_Enable_fb	D	HIGH = internal signal/noise generator is enabled and producing audio LOW = internal signal/noise generator is disable and silent
Generator_Frequency_fb#	A	**Generator_Frequency_fb#** represents the current internal generator's frequency parameter of 'Sine' type. The value is measured in Hz (0 Hz ~ 24000 Hz).
Generator_Mix_BUS_*[X]*_fb	D	**Generator_Mix_BUS_*[X]*_fb** show which DSP busses the internal signal/noise generator is currently mixed into. HIGH = the generator signal is mixed to BUS *[X]* at 0dB level LOW = the generator signal is not mixed to BUS *[X]*
GPO_*[X]*_fb	D	**GPO_[X]** represent the current state of the processor's GPO relays. HIGH = GPO [X] relays is closed LOW = GPO [X] relay is open
GPI_*[X]*_fb	D	**GPI_[X]** represent the current state of the processor's GPI digital inputs. HIGH = GPI [X] is HIGH (+5V / 50mA) LOW = GPI [X] is LOW (0V / 0mA)
Player_Play_fb	D	**Player_Play_fb** represents the current play/pause state of the processor's USB Media Player HIGH = Playing LOW = Paused
Player_Track_Name$	S	Full file name of the current track. Example: 'my_audio_file.wav'
Player_Tracks_Count#	A	Number of tracks available in the current Media Player folder.
Player_Current_Track_fb#	A	Index of the currently loaded track, among the available tracks of the current Media Player folder.

Output Signal Name	Type	Description
Player_Playback_Mode_fb	D	HIGH = File mode: playing a single track LOW = Folder mode: playing all tracks in the folder
Player_Repeat_Mode_fb	D	HIGH = Repeat enabled LOW = Repeat disabled
Player_Track_Time#	A	Current position of the loaded track playhead (in seconds)
Player_Track_Duration#	A	Total duration of the loaded track (in seconds)

PARAMETERS

Parameter Name	Type	Description
IpAddress	S	IP address of the Networked Audio Processor, for example "192.168.1.100". The IP address must be in the following ranges: - 10.0.0.1 – 10.255.255.254 (Class A) - 172.16.0.1 – 172.31.255.254 (Class B) - 192.168.0.1 – 192.168.255.254 (Class C) - 100.64.0.1 – 100.127.255.254 (SAS) - 169.254.0.1 – 169.254.255.254 (APIPA)
Meters_ANA_in **Meters_AES_in** **Meters_AVB_in** **Meters_MIC_in** **Meters_ANA_out** **Meters_AES_out** **Meters_AVB_out** **Meters_BUS** **Meters_MPL** **Meters_GEN**	A	These parameters are used to enable audio meters monitoring. - 0d = audio meters monitoring is disabled. All bits of **Meters_*[YYY]_[XX]#*** are LOW. - 1d = audio meters monitoring is enabled. Each bit of **Meters_*[YYY]_[XX]#*** gets HIGH if the associated channel/bus/generator audio level exceeds −60.0dBFS. **Important note** Enabling audio meters can be CPU-consuming for the CRESTRON processor, especially when multiple modules are inserted in the programs. They are disabled by default, and we recommend that they remain disabled unless this feature is absolutely necessary, or when the number of modules if less than 10. Alternatively, audio levels can be enabled only on a limited selection of modules. Please test your program first with audio levels disabled, and if CPU has good headroom when running the full programs, then try to enable signals and send normal audio to all unmuted networked audio processors to check that CPU is not going over 90%.

Partner: L-Acoustics
Model: P1
Device Type: Networked Audio Processor

TESTING

OPS USED FOR TESTING:	RMC3 v1.601.3857
SIMPL WINDOWS USED FOR TESTING:	4.14.20
CRESTRON DB USED FOR TESTING:	90.00.004.00
DEVICE DB USED FOR TESTING:	115.08.002.00
SAMPLE PROGRAM:	L-Acoustics Single Processor
REVISION HISTORY:	V. 1.0.0 First release V. 1.1.0 Added compatibility with firmware 2.10.x V. 3.2.0 Support of firmware 2.11.x *(version number aligned with Amplified Controller modules)*