



# BEATRICE R8

**8 Channel Network Audio Intercom**

**PRODUCT DETAILS**





# GlenSound Electronics Ltd

Thank you for choosing a new GlenSound product.

All rights reserved.

Information contained in this manual is subject to change without notice, if in doubt please contact us for the latest product information.

If you need any help with the product then we can be contacted at:

GlenSound Electronics Ltd  
The Design Buildings  
1 – 6 Brooks Place  
Maidstone  
Kent  
ME14 1HE  
United Kingdom

Telephone: +44 (0) 1622 753662

## EMAIL ADDRESSES

General enquires: [sales@glenSound.com](mailto:sales@glenSound.com)

Technical enquires: [techinfo@glenSound.com](mailto:techinfo@glenSound.com)

Sales enquires: [sales@glenSound.co.uk](mailto:sales@glenSound.co.uk)

## **PRODUCT WARRANTY:**

All equipment is fully tested before dispatch and carefully designed to provide you with trouble free use for many years.

We have a policy of supporting products for as long as possible and guarantee to be able to support your product for a minimum of 10 years.

For a period of one year after the goods have been despatched the Company will guarantee the goods against any defect developing after proper use providing such defects arise solely from faulty materials or workmanship and that the Customer shall return the goods to the Company's works or their local dealer.

All non-wear parts are guaranteed for 2 years after despatch and any defect developing after proper use from faulty materials or workmanship will be repaired under this warranty providing the Customer returns the goods to the Company's works or their local dealer.



This product can produce high sound levels via the headphone output.

Please take caution when operating this product as listening to excessively high peak or sustained levels of volume may permanently damage human hearing.




## EU DECLARATION OF CONFORMITY FOR:

### **GS-BEATRICE R8**

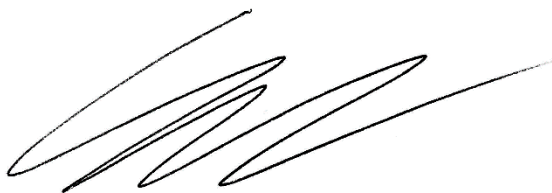
*8 Channel Dante/ AES67 desktop intercom*

This declaration of conformity is issued under the sole responsibility of the manufacturer.

This equipment is manufactured by Glensound Electronics Ltd of Brooks Place Maidstone Kent ME14 1HE is  marked and conforms to the following Union harmonisation legislation:

Low Voltage Directive:	EN60065 and EN62368-1:2014
Emissions:	BS EN55032:2015
Immunity:	BS EN55035:2017

Signed for and on behalf of Glensound Electronics Ltd.



Gavin Davis, Managing Director  
Maidstone, Kent, England

Date: 15/03/2018



## RoHS DIRECTIVE

EC directive 2002/95/EC restricts the use of the hazardous substances listed below in electrical and electronic equipment.

This product conforms to the above directive and for this purposes, the maximum concentration values of the restricted substances by weight in homogenous materials are:

Lead	0.1%
Mercury	0.1%
Hexavalent Chromium	0.1%
Polybrominated Biphenyls	0.1%
Polybrominated Diphenyl Ethers	0.1%
Cadmium	0.01%

# **WASTE ELECTRICAL AND ELECTRONIC EQUIPMENT REGULATIONS 2006 (WEEE)**

GlenSound Electronics Ltd is registered for business to business sales of WEEE in the UK our registration number is:

**WEE/JJ0074UR**

# **GLENSOUND BEATRICE R8**

## **Handbook Contents**

Issue 9

Description

Page No

### **Contents**

PRODUCT WARRANTY: .....	3
OVERVIEW .....	9
R8 FRONT PANEL LAYOUT .....	10
R8 REAR PANEL LAYOUT.....	13
MICROPHONE PHANTOM POWER .....	15
LOUDSPEAKER DIM LEVEL .....	16
SPEAK KEY CONFIGURATION.....	17
AUX/ IFB FUNCTION & DIMMING LEVEL.....	18
AUX/ IFB MONITORING FUNCTION .....	19
MON BUTTON CONFIGURATION.....	20
CALLING FEATURE .....	21
AUDIO BLOCK DIAGRAMS (analogue representation of digital circuits) .....	23
Audio Circuits TO Dante/ AES67 Network <i>SEE ALSO FIXED MIXER</i> .....	23
Audio Circuits FROM Dante/ AES67 Network <i>SEE ALSO FIXED MIXER</i> .....	24
FIXED MIXER .....	25
CONNECTING TO 4-WIRE EQUIPMENT.....	26
CONNECTING THE BEATRICE UNITS TO A DANTE NETWORK.....	26
Getting Dante Controller.....	26
Connecting Beatrice To The Network .....	26
Audio Over IP Network.....	26
Running Dante Controller .....	27
Dante Controller TIP .....	27
Device Not Showing Up In Dante Controller .....	28
AES67 MODE .....	29
UPDATING THE BROOKLYN MODULE.....	32
WIRING INFORMATION.....	33
XLR & JACK Wiring.....	33
D15 Socket Loop Input Wiring.....	35
D25 Plug Loop Output Wiring.....	36



## **OVERVIEW**

The BEATRICE range of IP intercoms was designed for broadcast, theatre and professional audio applications.

Our Beatrice intercom system utilises the reliable and proven Dante network audio transmission protocol to allow real time distribution of uncompressed audio across standard networks. As such the BEATRICE 8 channel units are also fully compatible with other manufacturers' equipment using the Dante protocol. The Beatrice R8 is also AES67 compliant.

All units in the system are designed to be very easy to use for the operator and simple to setup for the technician. They includes all the basic functionality required for small intercom systems and none of the overly complex installation requirements normally associated with large systems.

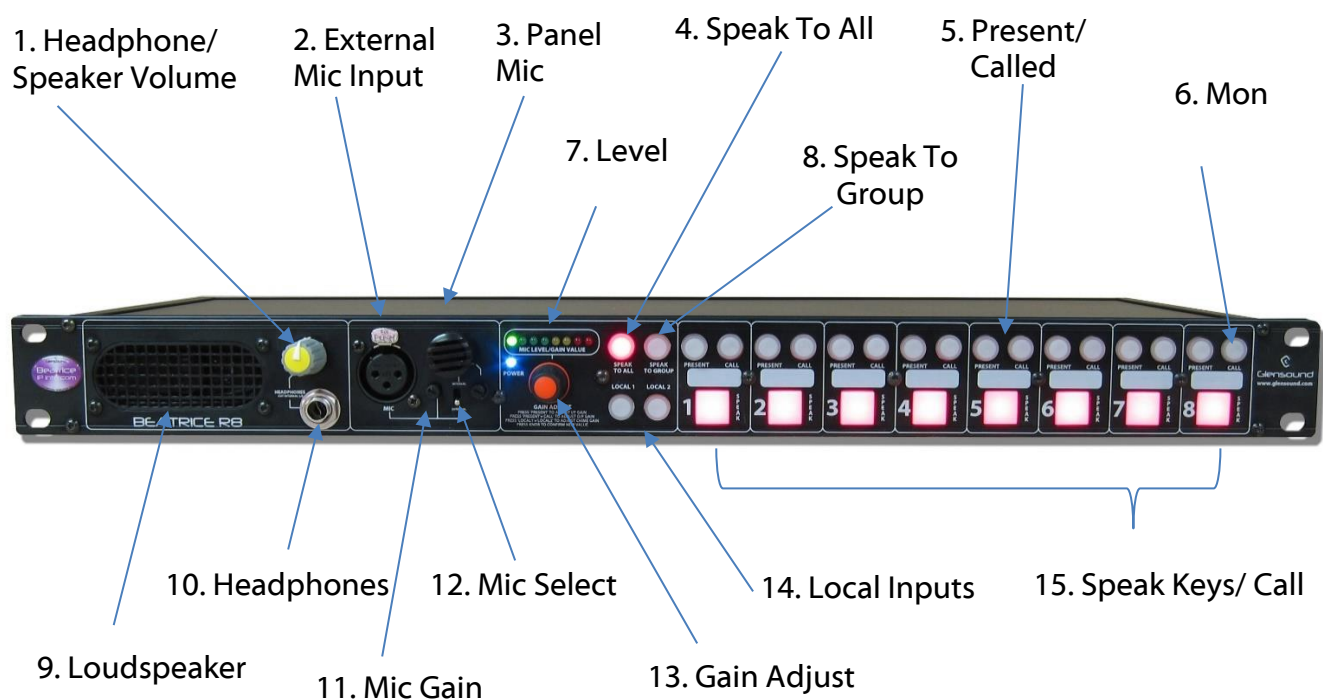
The Beatrice R8 provides 8 intercom circuits on and off the AoIP network. It also has 2 analogue inputs & outputs to the network that can be routed to allow connectivity to traditional 4-wire circuits. 8 off AoIP IFB/ AUX circuits are provided to allow versatility of system setup and an internal fixed ratio mixer provides even mor eflexibility from this simple but powerful subrack.

A front panel built in microphone and loudspeaker are provided, or alternatively, an external microphone (Gooseneck or headset) can be connected to the front panel XLR and headphones or external loudspeakers can also be used.

The name Beatrice was chosen for our intercom range as she was the love of Dante Alighieri:

*'Dante had fallen in love with another, Beatrice Portinari (known as Bice), whom he first met when he was only nine.'* Source Wikipedia.

## **R8 FRONT PANEL LAYOUT**



1.

### **2. Headphone / Speaker Volume**

This rotary knob adjusts the volume of the internal loudspeaker and front panel headphones.

### **3. External Microphone Input**

This is a balanced microphone input. It can accept dynamic and condenser microphones and phantom power (12v) can be turned on/ off via an internal link.

### **4. Panel Microphone**

The panel microphone is a small electret microphone fitted behind the rugged housing and is suitable for most communications purposes.

### **5. Speak to All Button**

Pressing this this button enables the operator to turn on and talk to all 8 outputs simultaneously.

### **6. Present/ Called Indicators**

There are 8 present indicators one for each talkback input from the audio network. These are both a LED indicator and a button.

When an incoming audio signal is detected on the associated incoming circuit the present LED will illuminate (threshold -20dB) to show that audio is being

received on that channel. The present LED will stay illuminated for 10 seconds after the incoming audio stops.

The present LED is also used to indicate an incoming call from another device. If an incoming call is detected then the present LED will flash

Pressing the present indicator button allows the audio input level of that circuit to be adjusted using the front panel Gain Adjust knob.

## 7. **Mon Button**

This yellow button is used to select incoming circuits to route to the monitor (loudspeaker/ headphones).

If the switch is on and illuminated then the source is currently being monitored.

For units shipped from the end of January 2020 onwards it is possible to select what happens when all the monitor buttons are turned off. See Mon Button Configuration on page 20 for further details.

## 8. **Level LEDs**

The level LEDs provide a number of functions.

- A) When the unit is first turned on The 2<sup>nd</sup> LED from the left will flash. This is a heartbeat signal indicating that internal processor is starting up. When it's started and ready the 2<sup>nd</sup> LED will stop flashing.
- B) The first LED from the left flashes if no network connection is present.
- C) When adjusting the input or output gain of one of the 8 channels or the input gain of the 2 local inputs then the LEDs are used to indicate the position of the volume control. With just 1 LED on on the left indicating a very low gain and all LEDs on indicating a very high gain. While in this mode the LEDs will be flashing to indicate that they are showing the gain position and not actual level.
- D) In all other circumstances the meter acts as a PPM to show the level being sent to the network from the selected microphone.

## 9. **Speak to Group**

The speak to group key is used to allow the operator to speak to a selection of outputs.

To make a group press and hold the Speak To Group key and simultaneously press a channel's talk key. Each time you press the channel's talk key the channel will be added/ subtracted from the group.

## 10. **Loudspeaker**

A front panel mounted loudspeaker for monitoring the incoming audio circuits. Note the loudspeaker is automatically turned off when a headphone jack is inserted into the headphone socket.

## 11. **Headphone Jack**

This standard 6.35mm (1/4") stereo headphone socket accepts headphones from 30 to 1000 Ohms.

## 12. **Mic Gain**

This rotary potentiometer adjusts the gain of the selected microphone.

## 13. **Mic Select**

This toggle switch selects which of the 2 microphones is to be used as the audio source. If the switch is in the up position then the panel mic is used and if the switch is in the down position the external microphone input is used.

## 14. **Gain Adjust**

This shaft encoder can be continuously rotated left and right and it also has a built in switch which is operated by pushing it.

It is used in conjunction with the Present, Present + Call or Local illuminated buttons to adjust the input (or output) gain of the associated channel.

When a channel has been selected to have its gain adjusted the Level LEDs will start to flash, showing the amount of gain currently being applied, where the first left LED indicates low gain and all the LEDs indicate maximum gain.

Once the desired amount of gain has been set then push the gain adjust shaft encoder to set this gain level.

## 15. **Local Inputs**

There are 2 'local' analogue inputs on the rear of the subrack. These are routed straight to the Dante/ AES67 network. Using the local input front panel button in conjunction with the gain adjust control allows the input level of these circuits to be adjusted.

## 16. **Speak/Call Key**

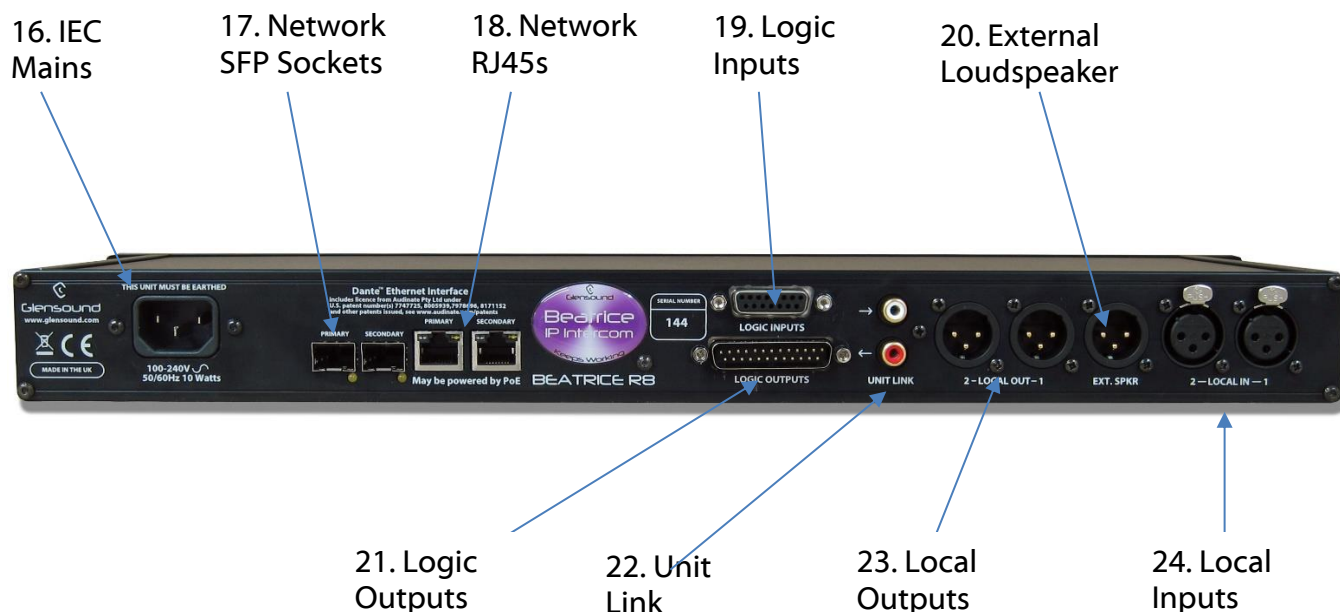
The 8 front panel speak keys allow the operator to talk to the 8 outgoing talkback circuits. On R8 units shipped from the beginning of January 2020 the operation of these talk keys can be configured by the user.

See Speak Key Configuration on page 17.

On units shipped before December 2019 the operation of the talk keys is fixed as intelligent lever key.

The speak keys also doubles as call switches, whereby a double tap of the speak key will send a call signal to the network.

## **R8 REAR PANEL LAYOUT**



### **17. IEC Mains**

The standard IEC mains plug accepts external AC voltages of 100 -240 VAC +/- 10%. There is an internal fuse.

### **18. Network SFPs**

There are 2 standard network data SFP sockets provided to allow redundant network connectivity if using Dante network audio protocol. They are both gigabit network connections. They accept standard SFP modules. The status LEDs flash to show when data is being correctly communicated with the attached switch.

If you require SFP modules then please email [sales@glen sound.com](mailto:sales@glen sound.com) for a quotation.

*Network Note (Applies to both 17 & 18)*

Factory default is to have the SFP & RJ45 sockets set to work in redundant mode. It is possible using Dante controller to set these network interfaces to work as a network switch instead of the redundant mode, meaning other network equipment can be connected to them.

### **19. Network RJ45s**

There are 2 standard network data RJ45 sockets provided to allow redundant network connectivity if using Dante network audio protocol. They are both gigabit network connections. The status LEDs flash to show when data is being correctly communicated with the attached switch.

Either of network sockets can be connected to a Power Over Ethernet (PoE) source to power the subrack. This PoE source can be a mid-span source or a network switch with PoE. If a PoE source is present on both circuits then this will act as a redundant power supply.

## 20. **Logic Inputs**

This 15 pin 'D' connector socket provides connectivity to external switches or devices for remote control of the talk keys and also provides inputs for cutting the internal or external loudspeakers.

For further details see 'WIRING' at the rear of this manual.

## 21. **External Loudspeaker**

This is a standard balanced analogue audio output for connection to external loudspeaker amplifiers. The output level is adjusted by the front panel Loudspeaker Volume control.

## 22. **Logic Outputs**

This 25 pin 'D' connector plug provides outputs connected to solid state relays that are activated when either any one of the speak keys is on (ideal for connecting to external LS dim circuits) or individually when any call received circuit is activated.

For further details see 'WIRING' at the rear of this manual.

## 23. **Unit Link**

These two RCA phono connectors carry S/PDIF digital audio with embedded data designed to be able to link two R8 units together, allowing microphone, loudspeaker and integrated talk circuits to be shared between two R8 intercom units.

To connect two R8 units together then two off RCA phono cables are required. Each cable should be connected between the white output RCA socket on one subrack and the red input RCA socket on the other rack.

Please note it is not possible to connect more than two R8 racks together.

## 24. **Local Outputs**

These two line level analogue audio outputs are fed directly from two incoming circuits from the Dante/ AES67 network.

## 25. **Local Inputs**

These two line level analogue inputs are routed via gain controls to two network audio output circuits.

## **MICROPHONE PHANTOM POWER**

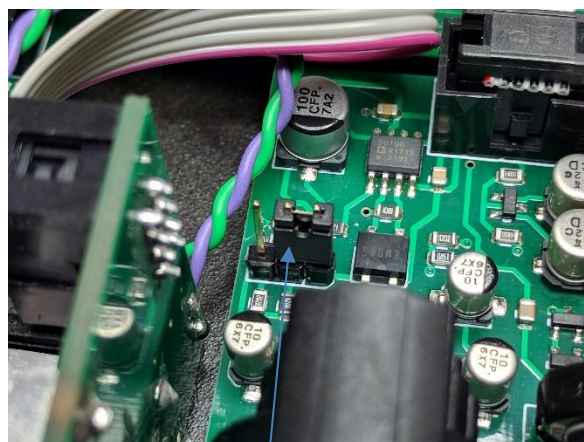
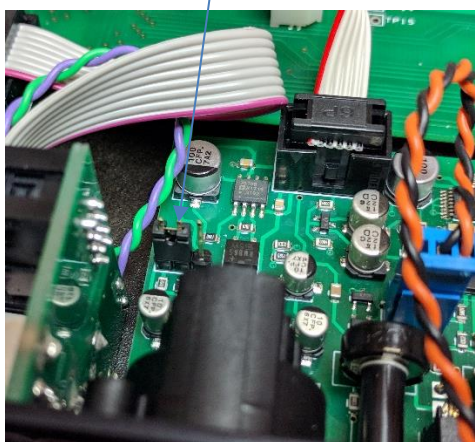
The factory default configuration of the R8 has the front panel microphone input set as +12v phantom power on. It is possible to de activate or activate +12v phantom panel by moving an internal jumper.

To turn on phantom power the lid of the unit must be removed.

**Please make sure the unit is turned off before starting this procedure**

To do this remove one of the sides of the unit, this is held in place by 4 screws (2 at the front & 2 at the rear). Then slide the lid out to the open side. You can then access the jumper shown in the 2 pictures below.

Jumper for Phantom Power OFF



Jumper for Phantom Power ON

After turning the phantom power on or off please replace the lid and re-fix the side panel.

## **LOUDSPEAKER DIM LEVEL**

The R8 has a loudspeaker dim circuit that automatically dims (reduces) the volume of the loudspeaker when any talkback key is operated.

During initial test at the factory the DIM amount will have been set to the factory default level. However it is possible to adjust this level should this be required.

On Beatrice R8 units shipped from the beginning of January 2020 enhanced firmware allows the DIM level to be set in software. Units prior to this date had an internal hardware control for setting the DIM level (if you need to change the DIM level of an older unit, please contact support for an older copy of this manual).

To access the LS DIM level configuration menu then please follow these instructions:

- 1) Start with the unit turned off (i.e. no power source connected)
- 2) Hold down SPEAK TO ALL **and** SPEAK TO GROUP button
- 3) While holding down the above two buttons apply power and turn the unit on
- 4) The two buttons can now be released and they will pulse to indicate that they are in LS DIM programming mode
- 5) Press the channel 1 speak key
- 6) The LEVEL LEDs will now indicate the currently set DIM level
- 7) The DIM LEVEL can be increased or decreased using the GAIN ADJUST control
- 8) Turning the GAIN ADJUST control anti-clockwise will reduce the amount of DIM that is applied when a speak key is pressed and turning it clockwise will increase the amount of DIM that is applied.
- 9) To leave the dim configuration menu just press either the SPEAK TO ALL or SPEAK TO GROUP button and the R8 will reset and the DIM level will be saved in non-volatile memory.



## **SPEAK KEY CONFIGURATION**

On R8 units shipped from the beginning of January 2020 it is possible to individually set the operation of the 8 speak keys.

### **Options**

There are 3 possible operational options for each talk key:

1) INTELLIGENT LEVER KEY: Whereby a brief press of the speak key toggles the circuit on/ off and a longer push/ hold of the speak key makes it act as Push To Talk (PTT).

*Whilst in configuration mode, Intelligent Lever Key is indicated by the four Green Level LEDs being on.*

2) PUSH TO TALK (PTT): The channel output will only ever be active while the speak key is pressed.

*Whilst in configuration mode, PTT is indicated by the two Yellow Level LEDs being on.*

3) LATCHING MODE: A brief press of the speak key toggles the circuit on/ off.

*Whilst in configuration mode, Latching Mode is indicated by the two Red Level LEDs being on.*

NOTE: Factory default is Intelligent Lever Key.

### **Configuration**

To access the Speak Key Configuration menu then please follow these instructions:

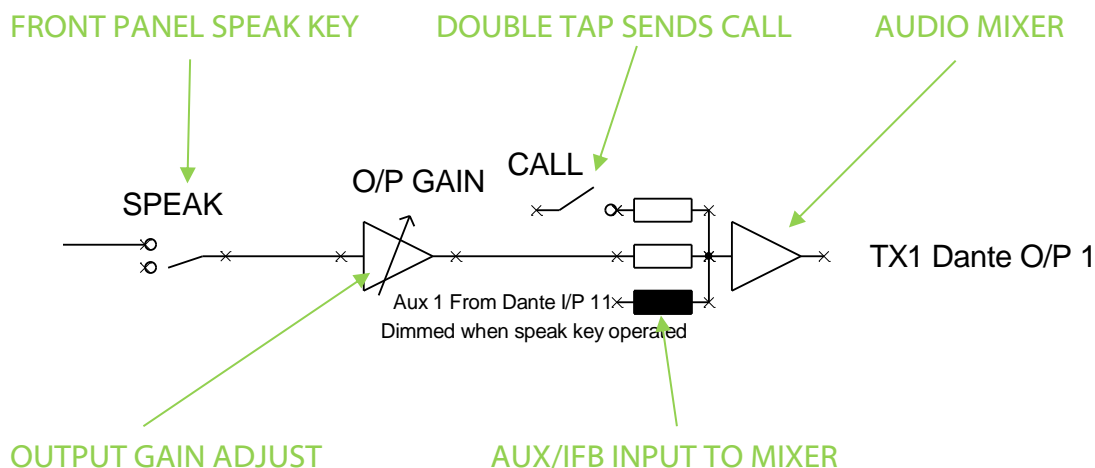
- 1) Start with the unit turned off (i.e. no power source connected)
- 2) Hold down SPEAK TO GROUP button
- 3) While holding down the Speak to group button apply power and turn the unit on
- 4) The button can now be released and it will pulse to indicate that you are in speak Key Configuration mode
- 5) Press and hold the channels speak key that you want to alter
- 6) While holding down a channels speak key use the Gain Adjust shaft encoder to select one of the three options as indicated on the Level LEDs (see options above)
- 6) To leave the Speak Key Configuration menu just press the SPEAK TO GROUP button and the R8 will reset and the configuration will be saved in non-volatile memory.

NOTE: Speak To All and Speak To Group buttons follow the configuration of channel 1.

NOTE: Speak Key Configuration does not alter the double tap to call feature.

## AUX/ IFB FUNCTION & DIMMING LEVEL

The audio block diagram below shows the audio workflow in one of the 8 outgoing talkback circuits.



The AUX/ IFB input is a very useful circuit and can be used for example to distribute programme audio to the user connected to the channels output, or maybe a clean feed circuit has been derived from mixing desk or router for onward sending to an outside source and this can be sent to that source via the AUX/ IFB input.

The AUX/ IFB Input is mixed into the channels outgoing talkback circuit. In units supplied prior to January 2020 then the AUX/ IFB input is automatically dimmed by 20dB whenever the associated channel's speak key is pressed.

In R8 racks supplied from January 2020 onwards then the amount of dimming that is applied to the AUX/ IFB input when the talk key is pressed can be set by the user for each channel.

### IFB Dimming Configuration

To access the IFB Dimming Configuration menu then please follow these instructions:

- 1) Start with the unit turned off (i.e. no power source connected)
- 2) Hold down LOCAL 1 button
- 3) While holding down the Local 1 button apply power and turn the unit on
- 4) The button can now be released and it will pulse to indicate that you are in speak IFB Dimming mode
- 5) Press and hold the channels speak key that you want to alter
- 6) While holding down a channels speak key use the Gain Adjust shaft encoder to adjust the dimming level from between -63dB and 0dB in 1dB steps. The Level Meter provides an indication of the amount of dim currently set.
- 6) To leave the IFB Dimming Configuration menu just press the Local 1 button and the R8 will reset and the configuration will be saved in non-volatile memory.

## AUX/ IFB MONITORING FUNCTION

The R8 has 8 audio inputs referred to on the block diagram as 'Dante I/P x' These 8 audio inputs are mixed together (assuming that their 'Mon' button is on) and sent to the loudspeaker/ headphone/ analogue and Dante LS outputs for monitoring.

The R8 also has 8 AUX/ IFB audio inputs that are mixed in with the associated channels' output (see page 18). These AUX/ IFB circuits by default are not monitored on any of the monitoring outputs.

In old fashioned analogue audio equivalent types of equipment to the R8 generally there would only be 8 audio inputs. These inputs would operate as both the AUX/ IFB and the monitoring input. Usually a front panel switch would route the input to the output (IFB mode) if required. This style of design meant that if you were using the unit as IFB then you would monitor the IFB audio input.

As the R8 has a separate AUX/ IFB input to the monitoring input then options exist as to how best make it operate in the same way as an old fashioned unit such that the user can monitor the Aux/ IFB feed.

### Option 1:

In the Dante network route the IFB signal to both the channels AUX/ IFB input and the channels I/P signal.

For example if you wanted to set this up for channel one, then in Dante controller you would route your incoming IFB audio to Dante I/P 1 and also Aux 1 on Dante input 11.

### Option 2:

It is possible to set each of the 8 channels individually to operate in 'IFB' mode. When set to IFB mode, then the monitoring chain is derived from the Aux/ IFB input and not from the Dante I/P x.

This means that any audio routed in the Dante network to a Dante I/P x input will not be monitored and will internally go nowhere within the R8. Whereas any audio that is routed to an AUX/IFB input will be both mixed in to the channels output (as an IFB) and also sent via the channels audio present LED and Monitoring button the audio monitoring circuits.

To configure the IFB mode for a channel then:

- 1) Start with the unit turned off (i.e. no power source connected)
- 2) Hold down SPEAK TO GROUP button **and the** LOCAL 1 button
- 3) While holding down both buttons apply power and turn the unit on
- 4) The buttons can now be released and they will pulse to indicate that you are in IFB configuration mode
- 5) Toggling a channels speak key will switch between IFB mode and normal operation for that channel. If the speak key is illuminated then IFB mode has been set for that channel.
- 6) To leave the IFB mode configuration just press the SPEAK TO GROUP button and the R8 will reset and the configuration will be saved in non-volatile memory.

## **MON BUTTON CONFIGURATION**

In units shipped from the end of January 2020 onwards it is possible to set what happens when all the Mon buttons are turned off.

There are two options:

- 1) When all Mon buttons are off then no incoming audio is monitored.
- 2) When all Mon buttons are off then a mix of all the incoming audio circuits will be monitored. (This is factory default)

To access Mon Button Configuration menu then please follow these instructions:

- 1) Start with the unit turned off (i.e. no power source connected)
- 2) Hold down SPEAK TO ALL button **and the** LOCAL 1 button
- 3) While holding down both buttons apply power and turn the unit on
- 4) The buttons can now be released and they will pulse to indicate that you are in Mon Button configuration mode
- 5) Speak 1 button toggles between the two Mon options.  
If Speak 1 button is illuminated then the monitoring circuit will act as per option 1 above.  
If Speak 1 button is not illuminated then the monitoring circuit will act as per option 2 above.
- 6) To leave the Call Masking setup menu just press the SPEAK TO ALL button and the R8 will reset, and the configuration will be saved in non-volatile memory.

## **SPEAK KEY GPO CONFIGURATION**

In units shipped from the end of August 2022 onwards it is possible to set the SPEAK KEY loop output operation.

There are two options:

- 1) When any of the speak keys are pressed the loop is active. (This is factory default)
- 2) When speak key 1 is pressed the loop is active.

To access SPEAK KEY loop configuration menu then please follow these instructions:

- 1) Start with the unit turned off (i.e. no power source connected)
- 2) Hold down SPEAK TO ALL button **and the** LOCAL 1 button
- 3) While holding down both buttons apply power and turn the unit on
- 4) The buttons can now be released, and they will pulse to indicate that you are in Mon Button configuration mode
- 5) Speak 3 button toggles between the two options.  
If Speak 3 button is illuminated, then the loop output will act as per option 2 above.

If Speak 3 button is not illuminated, then the loop output will act as per option 1 above.

- 6) To leave the setup menu just press the SPEAK TO ALL button and the R8 will reset, and the configuration will be saved in non-volatile memory.

## **D25 LOOP OUTPUTS - CALL OR INDIVIDUAL SPEAK KEY CONFIGURATION**

In units shipped from the end of August 2022 onwards it is possible to set the loop outputs to indicate either incoming call, or individual speak key operation.

There are two options:

- 1) Loop outputs indicate incoming call (This is factory default)
- 2) Loop outputs indicate individual speak key operation.

To access SPEAK KEY GPO Configuration menu then please follow these instructions:

- 1) Start with the unit turned off (i.e. no power source connected)
- 2) Hold down SPEAK TO ALL button **and the** LOCAL 1 button
- 3) While holding down both buttons apply power and turn the unit on
- 4) The buttons can now be released, and they will pulse to indicate that you are in Mon Button configuration mode
- 5) Speak 4 button toggles between the two Mon options.  
If Speak 4 button is illuminated, then the loop outputs will act as per option 2 above.  
If Speak 4 button is not illuminated, then the loop outputs will act as per option 1 above.
- 6) To leave the setup menu just press the SPEAK TO ALL button and the R8 will reset, and the configuration will be saved in non-volatile memory.

## **CALLING FEATURE**

### **General**

The Beatrice range of network audio intercoms include a calling feature. This can be used for alerting another user that you wish to communicate with them.

The calling feature works by sending an in band audio tone of 20kHz at -20dBFs for two seconds across the network to the device being called.

This audio tone is routed to the same destination as the audio output/ destination of the channel making the call.

The microcontrollers in Beatrice devices are constantly monitoring the incoming audio circuits of the incoming talkback channels for a 20kHz signal at -20dBFs, if it

detects this signal for 2 seconds then it will alert the user that an incoming call has been received.

### **Sending a Call**

To send a call on the Beatrice R8 just double tap the speak/call key of the intercom channel that you wish alert.

### **Receiving a Call**

When a call is received on a channel on the Beatrice R8 it will flash the round present illuminated button. This will continue to flash until the talk key is pressed.

An audible 'Chime' tone will also alert the user to an incoming call. In total there will be 6 'chimes' when a call is received. If the associated talk key is pressed then the chime signal will be muted.

### **Chime Level**

The audio level of the alerting chime that is played when an incoming call is detected can be adjusted by the user. To adjust this level press and hold both 'Local 1' and 'Local 2' buttons. While holding down the two buttons the Mic Level/ Gain Value LEDs will illuminate to indicate the current level. Turning the Gain Control shaft encoder clockwise will increase the chime level and turning it anti clockwise will decrease the chime level. Once the desired chime level has been set, press the Gain Control shaft encoder and then stop holding down the Local 1 and Local 2 buttons.

### **Call Masking**

On Beatrice R8 units shipped after the end of April 2019 enhanced firmware allows for the calling feature to be masked/ disabled on a channel by channel basis.

If you have an earlier Beatrice R8 then firmware will need to be updated to enable this facility. Unfortunately Beatrice R8 firmware can only be updated by Glensound, we are happy to update the firmware at our factory free of charge for you.

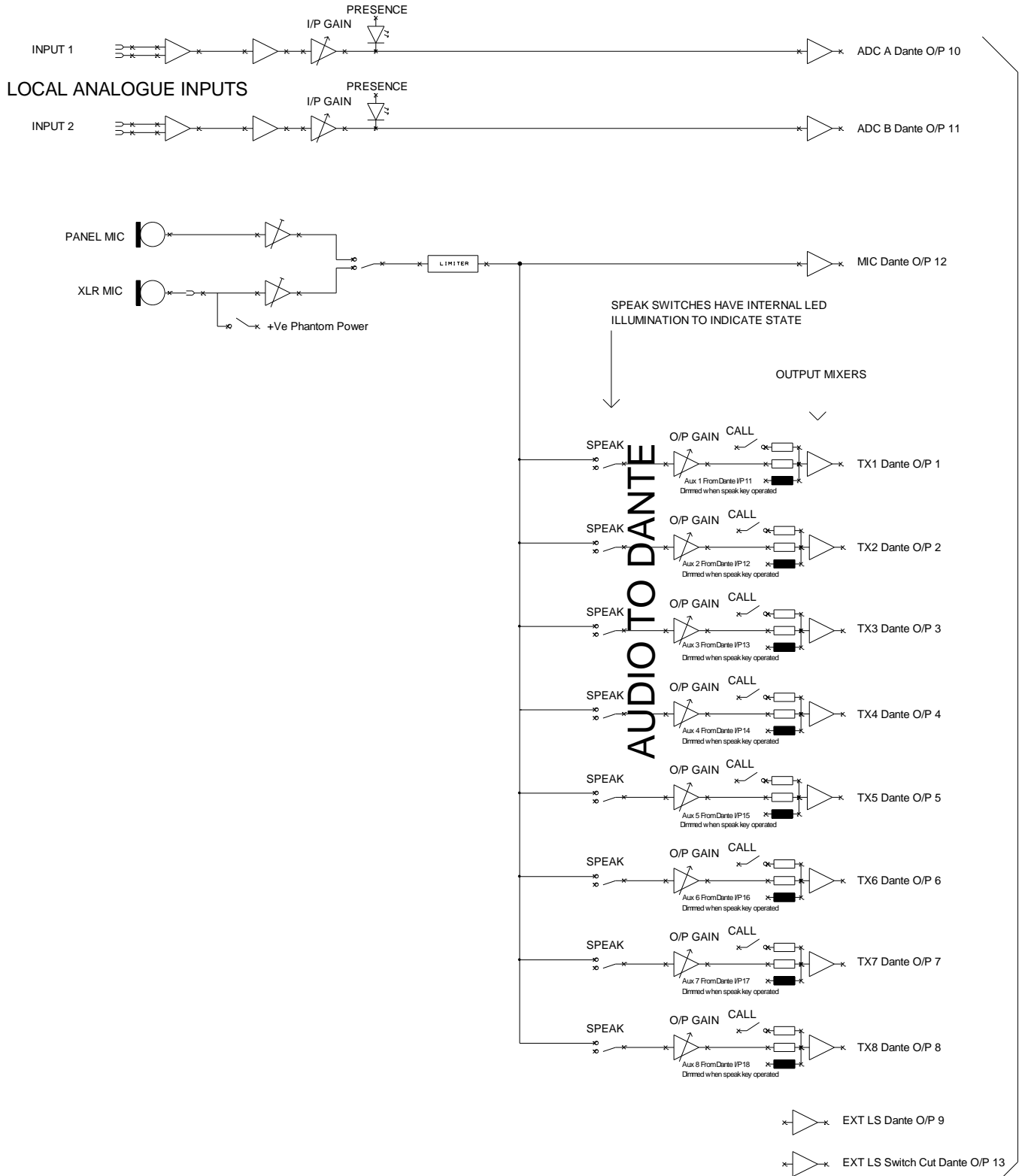
To access the Call Masking setup menu then please follow these instructions:

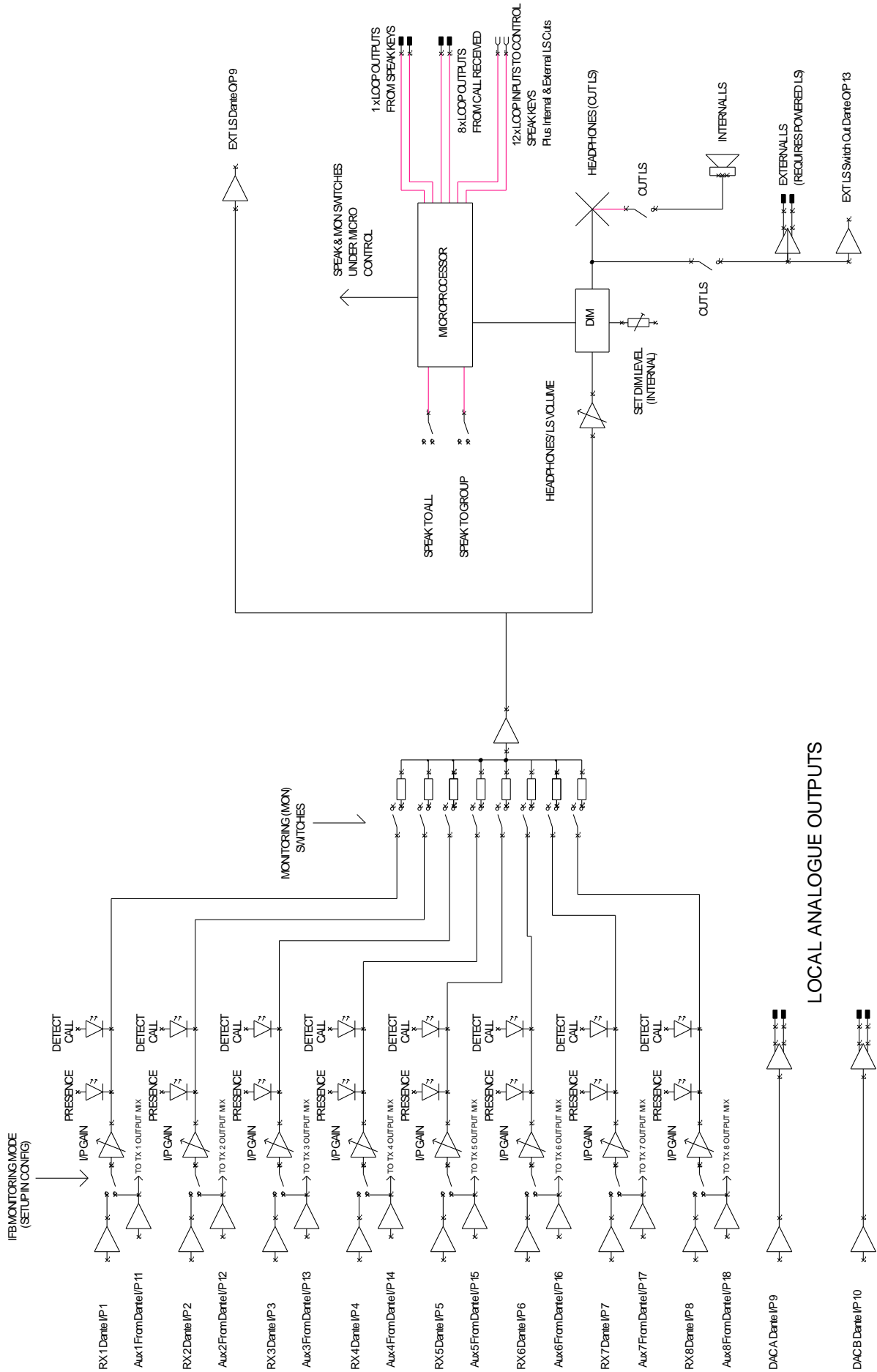
- 1) Start with the unit turned off (i.e. no power source connected)
- 2) Hold down SPEAK TO ALL button
- 3) While holding down the Speak to all button apply power and turn the unit on
- 4) The buttons can now be released and it will pulse to indicate that you are in Call Masking setup mode
- 5) In setup mode, the SPEAK buttons may be used to select channels for which CALL send and receive is activated. The default is for all channels to be enabled.
- 6) To leave the Call Masking setup menu just press the SPEAK TO ALL button and the R8 will reset and the configuration will be saved in non-volatile memory.

NOTE: Both double-tap-to-CALL and incoming call indication are disabled for the channels turned off in setup mode.

# AUDIO BLOCK DIAGRAMS (analogue representation of digital circuits)

## Audio Circuits TO Dante/ AES67 Network    *SEE ALSO FIXED MIXER*







## **FIXED MIXER**

The Beatrice R8 includes multiple fixed ratio mixers with inputs & outputs directly on/off the Dante/ AES67 network.

In a small simple intercom system there may be requirements for one user to listen to multiple sources on one single talkback channel. The internal fixed mixer within the Beatrice R8 is suitable for setting this up.

Groups are another possible requirement in small intercom systems whereby 2 or 3 operators (such as cameramen) are normally all on one single talkback circuit. The internal fixed mixer also helps with setting these circuits up.

The below table shows how these fixed ratio mixers are configured internally in the DSP. To use one or more of the internal mixers just route the audio to/ from the mixer channels in Dante controller.

Note that the default name in Dante Controller of the receivers are 'Mix In X' (whereby X is the channel number)

<b>MIX OUTPUT</b>		<b>SUM OFF</b>	<b>NOTES</b>
<i>Dante Output Channel No:</i>	<i>Default Name in Dante Controller:</i>	<i>Dante Receiver Channel Nos:</i>	
14	"Sum of 19 to 32"	19 to 32	
15	"Sum of 19 to 25"	19 to 25	
16	"Sum of 26 to 32"	26 to 32	
17	"Sum of 19 to 21"	19 to 21	
18	"Sum of 22 to 24"	22 to 24	
19	"Sum of 25 to 27"	25 to 27	
20	"Sum of 28 to 30"	28 to 30	
21	"Sum of 19 & 20"	19 and 20	
22	"Sum of 21 & 22"	21 and 22	
23	"Sum of 23 & 24"	23 and 24	
24	"Sum of 25 & 26"	25 and 26	
25	"Sum of 27 & 28"	27 and 28	
26	"Sum of 29 & 30"	29 and 30	
27	"Sum of 31 & 32"	31 and 32	
28	"Sum of 19 & 20 Dim"	19 and 20	Note Mix in 19 dimmed when signal present on Mix in 20
29	"Sum of 21 & 22 Dim"	21 and 22	Note Mix in 21 dimmed when signal present on Mix in 22
30	"Sum of 23 & 24 Dim"	23 and 24	Note Mix in 23 dimmed when signal present on Mix in 24
31	"Sum of 25 & 26 Dim"	25 and 26	Note Mix in 25 dimmed when signal present on Mix in 26
32	"Sum of 27 & 28 Dim"	27 and 28	Note Mix in 27 dimmed when signal present on Mix in 28

## **CONNECTING TO 4-WIRE EQUIPMENT**

The Beatrice R8 can be interfaced to a maximum of two traditional 4-wire systems.

To do this it is necessary to utilise the local audio inputs & local audio output circuits on the R8 and route these to/ from one of the talkback channels via the audio over IP network.

*Example to make Talkback Channel 8 operate as a 4 wire circuit.*

- A) In Dante controller route local input 1 (Dante transmitter no 10) to Dante receiver no 8 (the audio input to channel 8)
- B) In Dante controller route Talkback 8 output (Dante transmitter no 8) to Dante receiver no 9 (local output 1)

## **CONNECTING THE BEATRICE UNITS TO A DANTE NETWORK**

The Beatrice units are network audio devices utilizing the reliable and versatile Dante audio over IP protocol. Dante is a proprietary system (although very widely used) the originators of which are Audinate.

The information below is only meant as a very basic guide. Full details of the power of Dante network audio and instructions for using it can be found at [www.audinate.com](http://www.audinate.com)

### **Getting Dante Controller**

If you are connecting the Beatrice to a new Dante network the first thing you will need to do is to get the free Dante controller software from Audinate. This can be downloaded by visiting Audinate's web site at [www.audinate.com](http://www.audinate.com)

### **Connecting Beatrice To The Network**

Beatrice units can be connected to the network that you are going to use for your audio distribution simply by plugging in either, and, or any of the network connections on the rear. Once connected to the network it will be possible to see the Beatrice from within the Dante controller and route its' audio circuits.

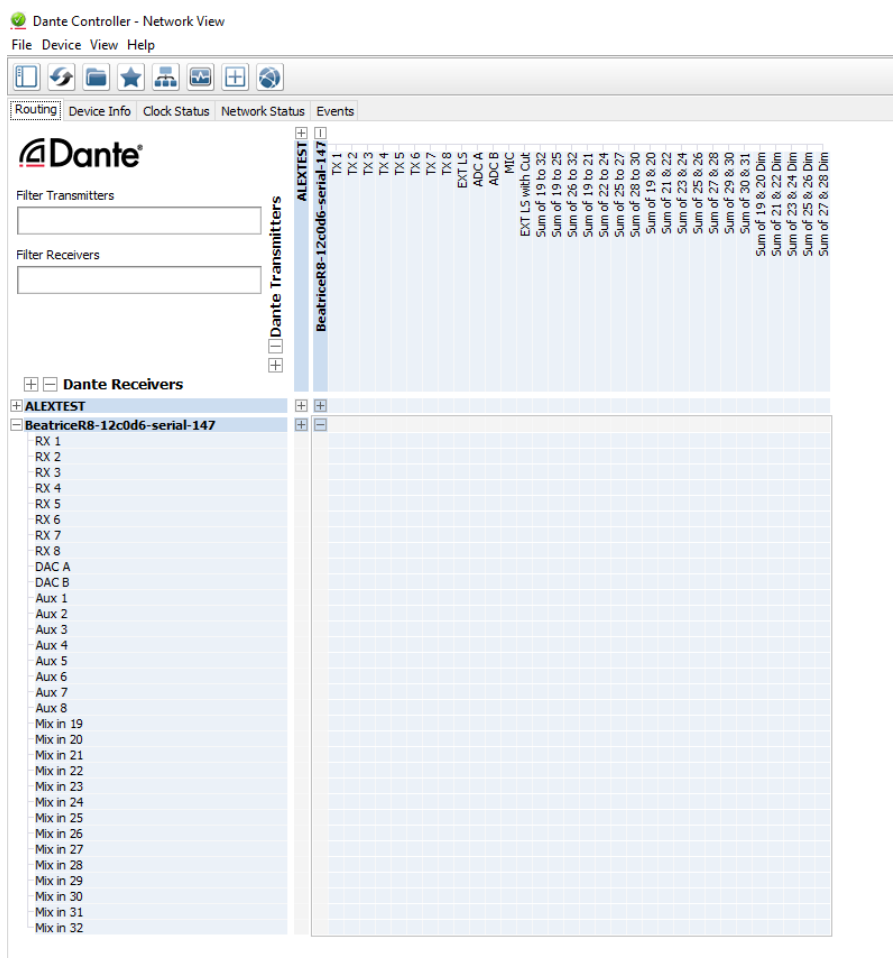
### **Audio Over IP Network**

We strongly recommend that you consider your network topology carefully and would not recommend sharing broadcast audio and general data on the same network.

For more details of audio over IP network structure please visit [www.audinate.com](http://www.audinate.com)

## Running Dante Controller

At the time of writing this manual the Dante Controller looks as per the screenshot below:



The Beatrice will have been named at the factory during test to allow them to be identified by the Dante controller.

The format used for the factory name is:

'BeatriceR8-snXXX'

Where 'BeatriceR8' refers to the Glensound product i.e. BeatriceR8.

The '-snXXX' refers to the serial number of the BeatriceD8 which can be found printed on the rear or side of the unit.

Note if you upload a new DNT file or clear the devices config then the name will change to BeatriceD8-xxXxXx whereby the 'X's refer to the devices MAC address.

### Dante Controller TIP

If you have never run Dante controller before then make sure that on the bottom left of the Dante controllers' screen 'P' or 'S' is next to a green square as this indicates that it is connected to a network. By clicking 'P' or 'S' a pop up box opens to allow you to set what network interface the controller is using.

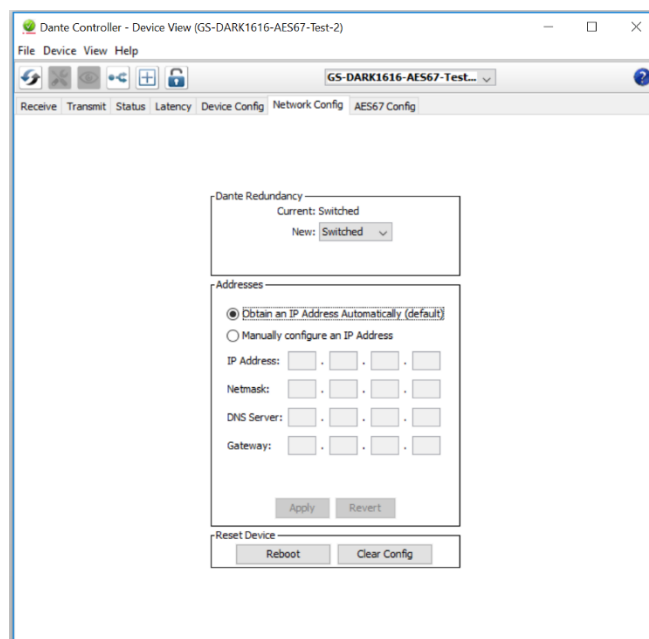
## Device Not Showing Up In Dante Controller

If your Dante™ device does not show up in Dante™ Controller then the most likely issue is that the device's IP Address is not appropriate for your network.

- A) It maybe that the device is set to obtain an IP address automatically using DHCP (this is the default configuration) and your network is setup for fixed IP addresses only and does not have a DHCP server.
- B) It maybe that the device has had a fixed IP address assigned but that this address is not suitable for your network.

The solution to both scenarios is basically the same.

- 1) You must connect your Dante™ device directly to the Ethernet port of your computer using an Ethernet cable.
- 2) Make sure that your computer is set to 'Obtain an IP address automatically'
- 3) After a few minutes the Dante™ device should now appear in Dante™ Controller.
- 4) Double click the device name to open up device view.
- 5) Open up the 'Network Config' tab
- 6) Either turn on 'Obtain an IP Address Automatically' or correctly configure the 'Manually configure an IP Address' options for your network.
- 7) Click on 'Apply' to confirm the new settings, then disconnect the computer and reconnect the Dante™ device to your network.



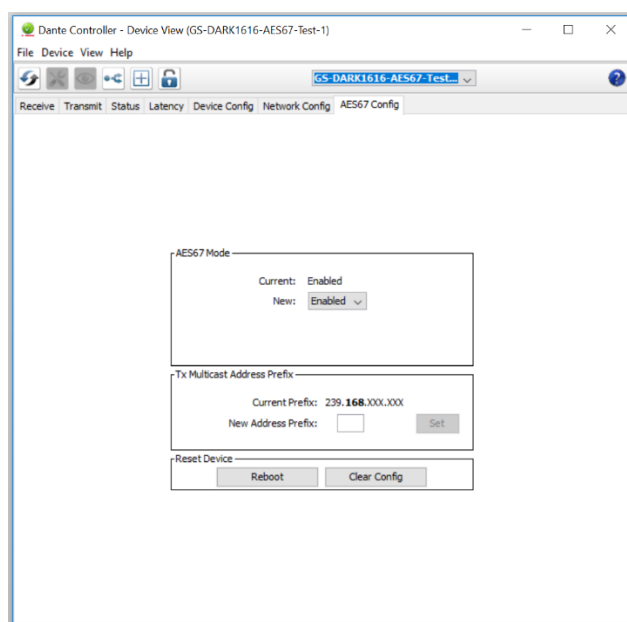
## **AES67 MODE**

The Beatrice uses a module from Audinate called a Brooklyn Module for its network audio interface. Audinate are the company behind Dante™ and as such the module's primary network audio protocol is Dante, however Audinate have enabled their module to comply with AES67 and therefore the Beatrice can be set to AES67 mode for interaction with other AES67 devices.

Please note however that Glensound are relying on Audinate's AES67 interface and are unfortunately not able to provide full AES67 support for the unit. AES67 support should be sought directly from Audinate.

### **1. Turning On AES67 Mode**


If you want to use your Beatrice on an AES67 network and it has not been set to AES67 mode then this can be set in Dante controller by double clicking the Paradiso to open the Device View window where you will find an AES67 tab to enable AES67 support.

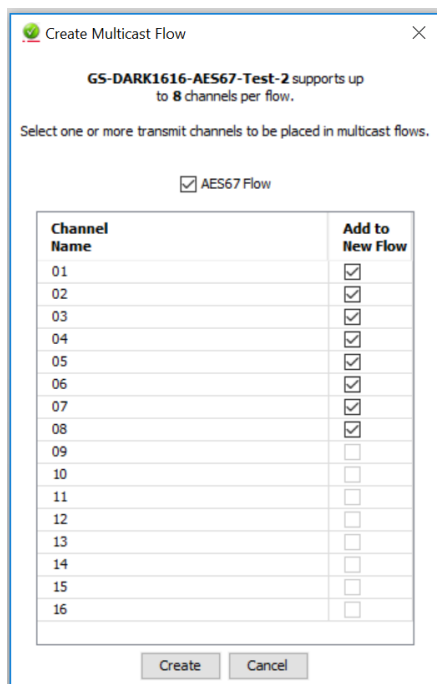


Once the AES67 drop down box has been enabled you'll have to reboot the Beatrice for the change to take effect. After the reboot go back to the AES67 tab and set the multicast prefix address to one that is suitable for your network.

## 2. Sending AES67 Audio

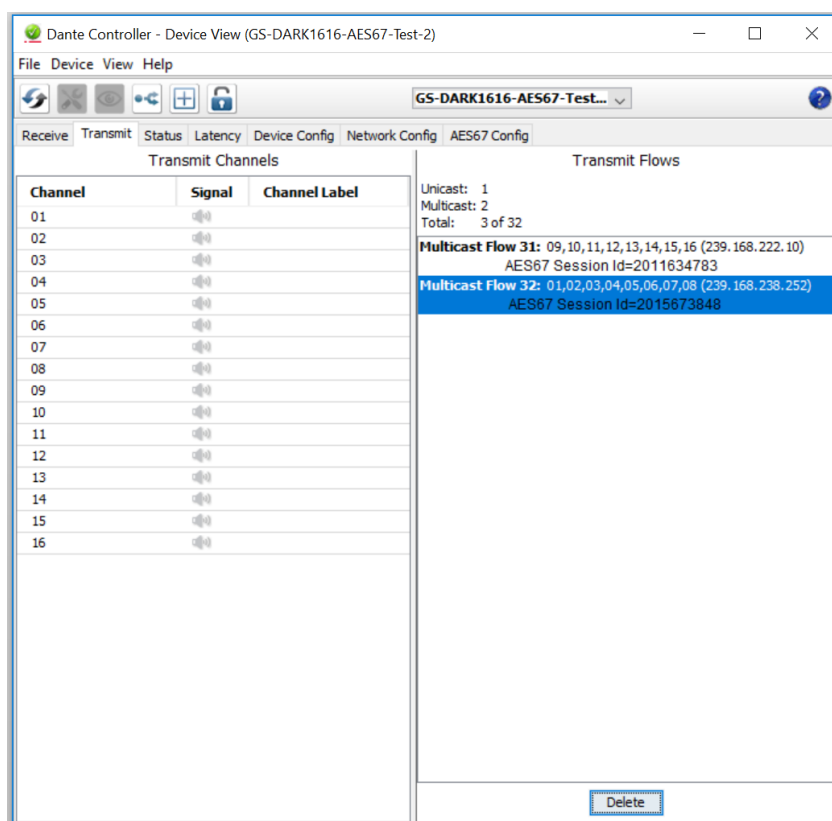
To transmit AES67 audio to the network a multicast flow must first be setup.

This is done by selecting the 'Create New Multicast Flow' Icon  in the Device View.



Tick the AES67 Flow check box, then select up to 8 channels to be included in the flow then click 'Create'

Once set the flows can be seen in the transmit tab of the device view.



### **3. Receiving AES67 Audio**

Once a compatible AES67 stream is detected on the network by Dante Controller the AES67 flows will appear in the Dante Transmitters section in the Routing tab.

### **4. AES67 Restrictions**

AES67 flows can only be generated with the following constraints:

- Multicast Only
- Non-redundant
- Destination address in range 239.nnn.0.0 to 239.nnn.255.255 (239.nnn/16), port 5004
- 48kHz sampling rate
- 24 bit linear (L24) encoding
- 1 msec packet time
- Up to 8 channels per stream

Received AES67 flows have the following constraints:

- Multicast Only
- Non-redundant
- Destination address in range 239.nnn.0.0 to 239.nnn.255.255 (239.nnn/16), port 5004. Must match destination address range.
- 48kHz sampling rate
- L16 or L24 encoding
- 125usec, 250usec, 333usec, 1 msec packet time
- Up to 8 channels per stream

## **UPDATING THE BROOKLYN MODULE**

The Brooklyn module is a device supplied by Audinate that does most of the processing for the actual Dante/ AES67 network audio streams. There is one Brooklyn module in each Beatrice R8. We supply special code (a .dnt file) that sets up/ initiates the Brooklyn module and makes it work in particular way and we also run extra code on its internal microprocessor to make it work correctly with the Beatrice D8.

### **1. Finding Out Current Installed Version**

Open Dante Controller.

Open Device Info tab.

Double Click in the device that you are working with....a new window will open called 'Device View (name of device)

Open the status tab.

The Firmware Version (of the Brooklyn module) will be found under the 'Device Information' heading.

### **2. Finding Out What The Latest Available Version Is**

Go the Beatrice R8's web page at <http://www.glen sound.co.uk/product-details/beatrice-r8/> and open the 'Firmware Latest Version' Tab.

This will give both the latest version numbers/ file names and also the location to download the file from.

### **3. Updating the Brooklyn Module**

The firmware that runs on the Brooklyn module is updated using Audinate's Firmware updating tool. The updating tool and a user guide can be downloaded from Audinate's website:

<https://www.audinate.com/products/firmware-update-manager>

The actual firmware to install using the above tool can currently be downloaded from:

[http://www.glen sound.co.uk/Software/Dante/BeatriceR8\\_v1.0.0\\_Dante\\_v4.0.7.8.dnt](http://www.glen sound.co.uk/Software/Dante/BeatriceR8_v1.0.0_Dante_v4.0.7.8.dnt)

When downloading the above file please make sure that you know where your computer has saved it.

Please note we strongly advise that when you do the update that only your PC and the Beatrice D8 that you want to update are on the network to save accidentally updating the wrong Dante device.

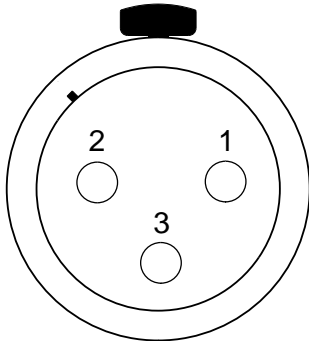
### **NOTE**

Once updated the devices name will change to BeatriceD8-xxXxXx whereby the 'X's refer to the MAC address. We recommend changing the name to something more convenient.



## WIRING INFORMATION

### XLR & JACK Wiring



XLR SOCKET (FEMALE)

#### STANDARD XLR AUDIO PINOUTS:

**1: Ground/ Earth**

**2: INPHASE/ POSITIVE/ MIC +**

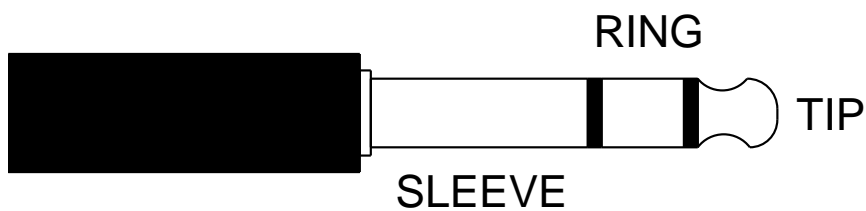
**3: MATE/ NEGATIVE/ MIC -**

#### STANDARD HEADPHONE WIRING:

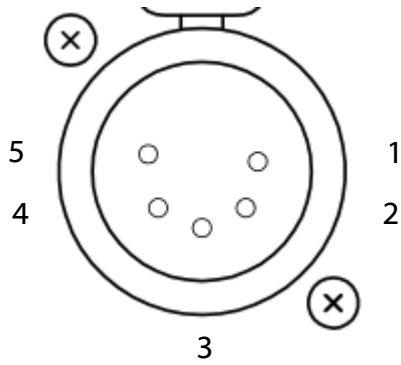
**TIP: A/ LEFT Ear**

**RING: B/ RIGHT Ear**

**SLEEVE: Common/ Earth**

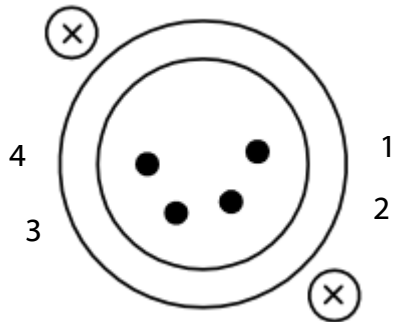


## 5 & 4 Pin Headset Wiring Options



### 5 PIN XLR AUDIO PINOUTS:

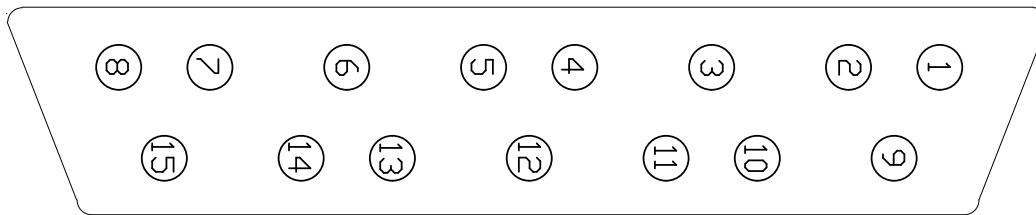
- 1: MATE/ NEGATIVE/ MIC -
- 2: INPHASE/ POSITIVE/ MIC +
- 3: GROUND/ EARTH
- 4: HEADPHONE LEFT
- 5: HEADPHONE RIGHT



### 4 PIN XLR AUDIO PINOUTS:

- 1: MIC GND/ MIC -
- 2: INPHASE/ POSITIVE/ MIC +
- 3: HEADPHONE GND
- 4: POSITIVE/ HEADPHONE +

## **D15 Socket Loop Input Wiring**

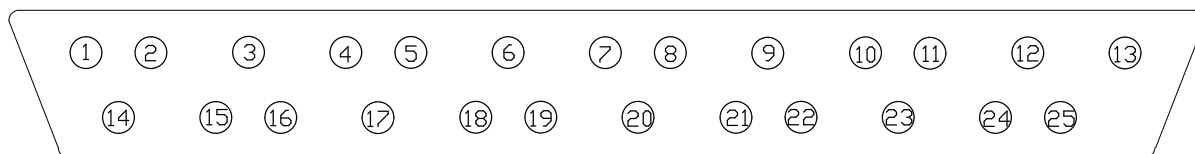


<b>PIN Number</b>	<b>Function</b>
1	Speak 1 Button
2	Speak 2 Button
3	Speak 3 Button
4	Speak 4 Button
5	Speak 5 Button
6	Speak 6 Button
7	Speak 7 Button
8	Speak 8 Button
9	Speak to Group
10	Speak to All
11	Cut Internal LS / HP Out
12	Cut EXT SPKR Out
13 - 14	Not connected
15	Ground/ Earth

The 12 active loop input pins work in Parallel with the front panel switches.

To operate an external loop just connect the input that you want to operate to the Ground/ Earth.

## **D25 Plug Loop Output Wiring**



<b>PIN Numbers</b>	<b>Function</b>
1 & 14	Call Received Channel 1
2 & 15	Call Received Channel 2
3 & 16	Call Received Channel 3
4 & 17	Call Received Channel 4
5 & 18	Call Received Channel 5
6 & 19	Call Received Channel 6
7 & 20	Call Received Channel 7
8 & 21	Call Received Channel 8
12 & 25	Speak Key On
13	Ground/ Earth
9, 10, 11, 22, 23 & 24	Not Connected

Loop outputs are connected internally to solid state relays which are wired normally open. Therefore the PIN Numbers in the above table refer to either side of the switched relay output that will become joined together if the associated output is active.

The maximum voltage handling for each relay is 400V.

The maximum current handling for each relay is 100mA.