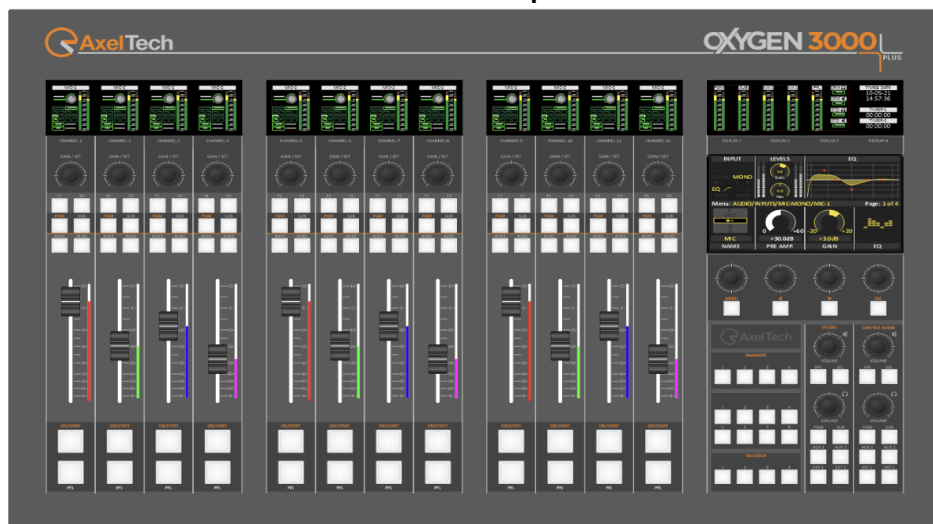


OXYGEN 3000 PLUS

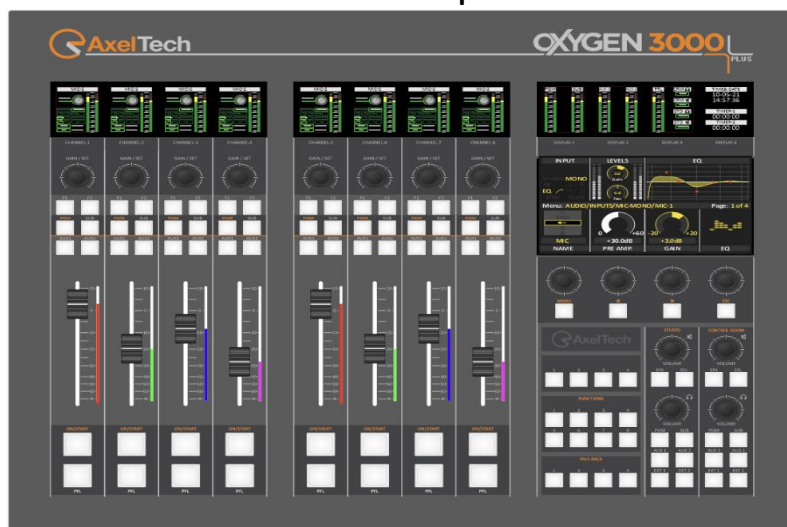
(Rev. 4.0-ENG)



OXYGEN 3000 PLUS | 16 FADERS



OXYGEN 3000 PLUS | 12 FADERS



OXYGEN 3000 PLUS | 8 FADERS

SUMMARY

SAFETY WARNINGS/ISTRUZIONI PER LA SICUREZZA	6
PREFACE.....	6
SAFETY WARNINGS.....	7
CONSIGNES DE SÉCURITÉ IMPORTANTES.....	9
ISTRUZIONI IMPORTANTI PER LA SICUREZZA	11
WICHTIGE SICHERHEITSHINWEISE.....	14
INSTRUCCIONES IMPORTANTES DE SEGURIDAD.....	16
UNPACKING AND INSPECTION	18
FIRST INSTALLATION RECOMMENDATIONS	19
POWER SUPPLY.....	19
INSTALLATION NOTE AND FIRST STEP	20
FIRST CONNECTION	21
INTRODUCTION	22
GENERAL DESCRIPTION	24
DIMENSIONS (8 PHYSICAL FADERS).....	24
DIMENSIONS (12 PHYSICAL FADERS).....	25
DIMENSIONS (16 PHYSICAL FADERS).....	26
TALKBOX DIMENSIONS	27
OUTPUT CONNECTIONS	30
SURFACE	33
CHANNEL CONTROLS.....	34
CHANNEL DISPLAY.....	35
GAIN/KNOB	35
F1 AND F2.....	36
PGM, SUB, AUX1 AND AUX2	36
FADER.....	37
FADER LED BAR	37
ON/START.....	38
PFL.....	39
SPECIAL FUNCTION BUTTONS	41
SNAPSHOTS	41
FUNCTIONS	42
TALKBACK.....	45
MONITOR SECTION.....	48
STUDIO & CONTROL ROOM	48
MENU CONTROL (BUTTONS & DISPLAY 7" TFT)	51
COLOR DISPLAY 7" - 800X480 RESOLUTION	51
BUTTONS FOR THE MENU NAVIGATION.....	52
ENCODER FOR THE MENU NAVIGATION.....	52
MENU DISPLAY PARTS.....	53
OUTPUTS DISPLAY	53
MENU	54
INPUT MODE CONFIGURATION	54
AUDIO	56
1.1 INPUT/OUTPUT MODE	56
AUDIO 1-2 INPUT	57

DANTE (INPUT/OUTPUT)	68
COMPUTER MODULE	74
1.2 SURFACE MODE	77
1.3 FADER SETTINGS	82
1.4 VJ PRO MODE	85
2. INPUTS	89
MIC (MIC-1 TO MIC-10)	89
MIC AUTOMIX	126
MONO	139
STEREO	156
USB AUDIO	164
DANTE* (OPTIONAL)	166
TELCO...	171
PHONE	180
BT	191
STREAMING	202
DIGITAL AES/EBU	211
VOIP	213
TALK SHOW	221
TONE GEN.	228
3.1 ANALOG OUTPUTS	233
ANALOG 1-2 — XLR	234
ANALOG 3-4 — RJ45	239
HEADPHONES 1	240
3.2 DIGITAL OUTPUTS	241
DIGITAL 1 (AES/EBU)	241
USB 2	247
STREAMER 1	251
3.3 DANTE OUTPUT* (OPTIONAL)	254
DANTE INPUT/OUTPUT MODE	254
3.4 MONITOR OUTPUT	260
SPK-CRM	260
HDP-CRM	263
SPK-STUDIO	265
HDP-STUDIO	268
SETTINGS	271
SNAPSHOT	279
GENERAL SETTINGS	287
1. GPIO – GENERAL-PURPOSE INPUTS AND OUTPUTS	287
GPI-1 BOARD	288
GPI-2 BOARD	292
GPO-1 BOARD	293
GPO-2 BOARD	297
MIC TALKBOX	297
2. COMMUNICATIONS	300
TCP-IP...	300
TIME&DATE	304
3. ACCESS CODE	307
4. LIGHT & DISPLAY	308
BUTTON LIGHT:	308
DISPLAY	314

HDMI LAYOUT	316
ENERGY SAVING	318
5. MR LIGHT	322
MRL-CR1 / MRL-CR2	322
MRL-ST1 / MRL-ST2	328
MRL 1-8	329
6. DMX	331
DMX-CR1 / DMX-CR2	331
DMX-ST1 / DMX-ST2	337
DMX 1-8	338
7. SMART KEYS	339
BUTTON COLOR	339
K1-K8...	340
8. ONAIR TRIGGERS	342
SERVICE	345
1. CONFIGURATION	345
FACTORY RESET	346
SAVE.....	347
RESTORE	348
AUTOSAVE INTERVAL	349
2. SOFTWARE	350
UPGRADE	350
VERSION	354
WATCHDOG	355
3. LOGS	356
SAVE TO USB	356
SYSTEM LOGS	357
4. WEB LOGIN	358
PASSWORD	358
5. DIAGNOSTIC	360
DIAGNOSTIC LOGS	360
RESET LOGS	361
RESET SURFACE	362
PSU SUPERVISOR	362
TEST CONFIGURATION	363
START SURFACE TEST	366
6. SURFACE LOCK	366
7. KICK REMOTE CLIENT	367
8. PERFORMANCE	368
PULSAR	368
DISPLAY REFRESH	369
DASHBOARD MODE	369
DASHBOARD REFRESH	370
9. DETECT HDMI DISPLAY	371
10. RECOVERY	371
CREATE RESTORE POINT	372
FORMAT SD	373
SAFE MODE – SWITCH OFF	374
REBOOT	375
+ 187-RJ45-MIC	376
+ 188-RJ45-TELCO	377

+ 189-RJ45-LINE-IN	378
+ 190-RJ45-LINE-OUT	379
TECH SPECS	380
WEEE DIRECTIVE – INFORMATIVA RAEE	384
WARRANTY	385

SAFETY WARNINGS/ISTRUZIONI PER LA SICUREZZA

SAFETY WARNINGS

CONSIGNES DE SÉCURITÉ IMPORTANTES

ISTRUZIONI IMPORTANTI PER LA SICUREZZA

WICHTIGE SICHERHEITSHINWEISE

INSTRUCCIONES IMPORTANTES DE SEGURIDAD

(Rel. 6.2.0.2)

PREFACE

For your safety and to prevent the warranty from being accidentally invalidated, please read carefully all the texts marked with the Warning Symbols



The information contained in this manual is subject to change without notice and does not constitute a commitment by the seller.

The manufacturer will not be liable for any loss or damage resulting from the use of information or any errors contained in this manual or resulting from any erroneous operation or hardware failure contained in the product.

It is recommended that any repair and maintenance of the product be carried out by the manufacturer or its authorized agents. The manufacturer assumes no responsibility for any loss or damage caused by service, maintenance, or repair by unauthorized personnel.

SAFETY WARNINGS

The installation and servicing instructions in this manual are for use by qualified personnel only.

Read All Instructions. All safety and operating instructions must be read before operating the product. They also must be retained for future reference, as it contains many useful hints for determining the best combination of equipment settings for Yr particular application.

Heed All Warnings. All warnings on the product and those listed in the operating instructions must be adhered to.

Heat. This product must be situated away from any heat sources such as radiators or other products (including power amplifiers or transmitters) that produce heat.

Power Sources. This product must be operated from the type of power source indicated on the marking label and in the installation instructions. If you are not sure of the type of power supplied to your facility, consult your local power company. Make sure the AC main voltage corresponds to that indicated in the technical specifications. If a different voltage (ex. 110/115 VAC) is available, open the equipment closure and set the voltage switch on the main supply circuit, located behind the AC socket.

Power Cord Protection. Power supply cords must be routed so that they are not likely to be walked on or pinched by items placed upon or against them. Pay particular attention to the cords at AC wall plugs and convenience receptacles, and at the point where the cord plugs into the product.

Use only with a cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.

Lightning. For added protection for this product during a lightning storm, or when it is left unattended and unused for long periods, unplug it from the AC wall outlet and the audio connections. This will prevent damage to the product due to lightning and power-line surges.

Installation. Configuration and installation should only be carried out by a competent installation engineer.

Cabling. Using high-quality wires, well-protected. Make sure the cable is integrity.



This symbol alerts you to the presence of dangerous voltage inside the closure – voltage that may be sufficient to constitute a risk of shock. Do not perform any servicing other than that contained in the operating instructions. Refer all servicing to qualified personnel.



The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.



Do not change the voltage setting or replace the mains fuse without first turning the unit off and unplugging the mains cord.



Make sure the AC main voltage corresponds to that indicated in the technical specifications.

THIS APPARATUS MUST BE EARTHED!



To avoid the risk of fire, use the correct value fuse, as indicated on the label stuck on the right side of the unit.



This apparatus uses a single-pole main switch and does therefore not separate the unit completely from the main power. To completely separate from mains power (f.i. in the event of danger) unplug the mains power cord. As the MAINS plug is the disconnect device, the disconnect device shall remain readily operable.

CONSIGNES DE SÉCURITÉ IMPORTANTES

Lire ces consignes.

Conserver ces consignes.

Observer tous les avertissements.

Suivre toutes les consignes.

Ne pas utiliser cet appareil à proximité de l'eau.

Ne pas obstruer les ouvertures de ventilation. Installer en respectant les consignes du fabricant.

Ne pas installer à proximité d'une source de chaleur telle qu'un radiateur, une bouche de chaleur, un poêle ou d'autres appareils (dont les amplificateurs) produisant de la chaleur.

Ne pas annuler la sécurité de la fiche de terre, la troisième branche est destinée à la sécurité. Si la fiche fournie ne s'adapte pas à la prise électrique, demander à un électricien de remplacer la prise hors normes.

Protéger le cordon d'alimentation afin que personne ne marche dessus et que rien ne le pince, en particulier aux fiches, aux prises de courant et au point de sortie de l'appareil.

Utiliser uniquement les accessoires spécifiés par le fabricant.

Utiliser uniquement avec un chariot, un support ou une table spécifié par le fabricant ou vendu avec l'appareil. Si un chariot est utilisé, déplacer l'ensemble chariot–appareil avec précaution afin de ne pas le renverser, ce qui pourrait entraîner des blessures.

Débrancher l'appareil pendant les orages ou quand il ne sera pas utilisé pendant longtemps.

Confier toute réparation à du personnel qualifié. Des réparations sont nécessaires si l'appareil est endommagé d'une façon quelconque, par exemple: cordon ou prise d'alimentation endommagé, liquide renversé ou objet tombé à l'intérieur de l'appareil, exposition de l'appareil à la pluie ou à l'humidité, appareil qui ne marche pas normalement ou que l'on a fait tomber.

NE PAS exposer cet appareil aux égouttures et aux éclaboussures. Ne pas poser des objets contenant de l'eau, comme des vases, sur l'appareil.



Ce symbole indique la présence d'une tension dangereuse dans l'appareil constituant un risque de choc électrique.



Ce symbole indique que la documentation fournie avec l'appareil contient des instructions d'utilisation et d'entretien importantes.



Avant de modifier le commutateur de changement de tension ou remplacer le fusible il faut débrancher l'appareil de la prise électrique. Pendant son usage, l'appareil doit être branché à la prise de terre.



Utiliser le fusible principal AC avec la valeur qui est indiquée sur l'étiquette collée sur le coffret.



Assurez-vous que la tension principale AC correspond à celle indiquée dans les spécifications techniques.



L'interrupteur d'alimentation interrompt un pôle du réseau d'alimentation excepté le conducteur de terre de protection. En cas de danger, débrancher le cordon d'alimentation. Parce que la prise du réseau de alimentation est utilisée comme dispositif de déconnexion, ce dispositif doit demeurer aisément accessible.

ISTRUZIONI IMPORTANTI PER LA SICUREZZA

Leggere le presenti istruzioni.

Conservare queste istruzioni.

Osservare tutte le avvertenze.

Seguire scrupolosamente tutte le istruzioni.

Non usare questo apparecchio in prossimità di acqua.

Non ostruire alcuna apertura per il raffreddamento. Installare l'apparecchio seguendo le istruzioni.

Non installare l'apparecchio accanto a fonti di calore quali radiatori, aperture per l'afflusso di aria calda, forni o altri apparecchi (amplificatori inclusi) che generino calore.

Non rimuovere il terminale di connessione a terra sul cordone di alimentazione: esso ha lo scopo di tutelare l'incolumità dell'utilizzatore. Se la spina in dotazione non si adatta alla presa di corrente, rivolgersi ad un elettricista per far eseguire le modifiche necessarie.

Evitare di calpestare il cavo di alimentazione o di comprimerlo, specialmente in corrispondenza della spina e del punto di inserzione sull'apparato.

Utilizzare solo dispositivi di collegamento e gli accessori specificati dal produttore.

Utilizzare l'apparecchio solo con un carrello, un sostegno, una staffa o un tavolo di tipo specificato dal produttore o venduto insieme all'apparecchio. Se si utilizza un carrello, fare attenzione negli spostamenti per evitare infortuni causati da ribaltamenti del carrello stesso.

Scollegare l'apparecchio dalla presa di corrente durante i temporali o quando inutilizzato a lungo.

Per qualsiasi intervento, rivolgersi a personale di assistenza qualificato. È necessario intervenire sull'apparecchio ogniqualvolta si verificano danneggiamenti di qualsiasi natura. Ad esempio, la spina o il cavo di alimentazione sono danneggiati, è entrato liquido nell'apparecchio o sono caduti oggetti su di esso, l'apparecchio è stato esposto alla pioggia o all'umidità, non funziona normalmente o è caduto.

Non esporre a sgocciolamenti o spruzzi. Non appoggiare sull'apparecchio oggetti pieni di liquidi, ad esempio vasi da fiori.

Il prodotto deve essere connesso ad impianti costruiti secondo la regola dell'arte e muniti di protezione differenziale del circuito con valore non superiore agli 0,03A.

Tenere il prodotto lontano da liquidi.

Il prodotto deve essere utilizzato solo se integro e non danneggiato. Se il prodotto è stato sottoposto a forti urti o fosse venuto a contatto con liquidi è necessario contattare l'assistenza prima di accenderlo.

Il prodotto non v  aperto per nessun motivo, non v  modificato o manomesso. E' vietato tentare qualsiasi tipo di riparazione.

E' obbligatorio leggere il manuale utente prima di utilizzare il prodotto.

Il prodotto deve essere utilizzato da persone adulte. Tenere il prodotto fuori dalla portata dei bambini

Il prodotto v  collegato ad impianti costruiti secondo la regola dell'arte e muniti di protezioni magnetotermiche del circuito.

E' proibito sovraccaricare le prese di corrente. E' obbligatorio spegnere il prodotto se non utilizzato.

E' proibito ostruire le aperture di raffreddamento e aerazione.

E' obbligatorio tenere materiali infiammabili/combustibili lontani dal prodotto.

E' vietato utilizzare il prodotto in presenza di sostanze che possano creare atmosfera esplosiva.

Il prodotto v  utilizzato posizionato e utilizzato in maniera stabile.



Questo simbolo indica la presenza di alta tensione all'interno dell'apparecchio, che comporta rischi di scossa elettrica.



Questo simbolo indica la presenza di istruzioni importanti per l'uso e la manutenzione nella documentazione in dotazione all'apparecchio.



Non sostituire il fusibile o cambiare la tensione di alimentazione senza aver prima scollegato il cordone di alimentazione. **L'APPARATO DEVE ESSERE CONNESSO A TERRA.**



Sostituire il fusibile generale con uno di identico valore, come indicato sulla etichetta applicata sul mobile dell'apparato



Assicurarsi che la tensione di rete corrisponda a quella per la quale è configurato l'apparecchio.



Questo apparato utilizza un interruttore di alimentazione di tipo unipolare e l'isolamento dalla rete elettrica non è pertanto completo. Per ottenere un isolamento totale (ad esempio in caso di pericolo), scollegare il cordone di alimentazione. Inoltre, poichè la spina di alimentazione è utilizzata come dispositivo di sezionamento, essa deve restare facilmente raggiungibile.

WICHTIGE SICHERHEITSHINWEISE

Diese Hinweise LESEN.

Diese Hinweise AUFHEBEN.

Alle Warnhinweise BEACHTEN.

Alle Anweisungen BEFOLGEN.

Dieses Gerät NICHT in der Nähe von Wasser verwenden.

KEINE Lüftungsöffnungen verdecken. Gemäß den Anweisungen des Herstellers einbauen.

Nicht in der Nähe von Wärmequellen, wie Heizkörpern, Raumheizungen, Herden oder anderen Geräten (einschließlich Verstärkern) installieren, die Wärme erzeugen.

Die Schutzfunktion des Schukosteckers NICHT umgehen. Bei Steckern für die USA gibt es polarisierte Stecker, bei denen ein Leiter breiter als der andere ist; US-Stecker mit Erdung verfügen über einen dritten Schutzleiter. Bei diesen Steckerausführungen dient der breitere Leiter bzw. der Schutzleiter Ihrer Sicherheit. Wenn der mitgelieferte Stecker nicht in die Steckdose passt, einen Elektriker mit dem Austauschen der veralteten Steckdose beauftragen.

VERHINDERN, dass das Netzkabel gequetscht oder darauf getreten wird, insbesondere im Bereich der Stecker, Netzsteckdosen und an der Austrittsstelle vom Gerät.

NUR das vom Hersteller angegebene Zubehör und entsprechende Zusatzgeräte verwenden.

NUR in Verbindung mit einem vom Hersteller angegebenen oder mit dem Gerät verkauften Transportwagen, Stand, Stativ, Träger oder Tisch verwenden. Wenn ein Transportwagen verwendet wird, beim Verschieben der Transportwagen-Geräte- Einheit vorsichtig vorgehen, um Verletzungen durch Umkippen.

Das Netzkabel dieses Geräts während Gewittern oder bei längeren Stillstandszeiten aus der Steckdose ABZIEHEN.

Alle Reparatur- und Wartungsarbeiten von qualifiziertem Kundendienstpersonal DURCHFÜHREN LASSEN. Kundendienst ist erforderlich, wenn das Gerät auf irgendeine Weise beschädigt wurde, z.B. wenn das Netzkabel oder der Netzstecker beschädigt wurden, wenn Flüssigkeiten in das Gerät verschüttet wurden oder Fremdkörper hineinfließen, wenn das Gerät Regen oder Feuchtigkeit ausgesetzt war, nicht normal funktioniert oder fallen gelassen wurde.

Dieses Gerät vor Tropf- und Spritzwasser SCHÜTZEN. KEINE mit Wasser gefüllten Gegenstände wie zum Beispiel Vasen auf das Gerät STELLEN.



Dieses Symbol zeigt an, dass gefährliche Spannungswerte, die ein Stromschlagrisiko darstellen, innerhalb dieses Geräts auftreten.



Dieses Symbol zeigt an, dass das diesem Gerät beiliegende Handbuch wichtige Betriebs- und Wartungsanweisungen enthält.



Vor Änderung der Netzspannung oder Sicherungswechsel Netzkabel trennen. Das Gerät muss für den Betrieb geerdet werden.



Vor Änderung der Netzspannung oder Sicherungswechsel Netzkabel trennen. Das Gerät muss für den Betrieb geerdet werden.



Hauptsicherung nur mit einer gleichwertigen austauschen (s. entsprechende Etikette).



Vor Einschalten Netzspannungseinstellung am Gerät überprüfen bzw. anpassen.



Inpoliger Netzschalter. In Notfälle oder für Wartungsarbeiten Netzkabel trennen. Der Netzstecker fungiert auch als Trennelement muss deshalb zugänglich bleiben.

INSTRUCCIONES IMPORTANTES DE SEGURIDAD

LEA estas instrucciones.

CONSERVE estas instrucciones.

PRESTE ATENCION a todas las advertencias.

SIGA todas las instrucciones.

NO utilice este aparato cerca del agua.

NO obstruya ninguna de las aberturas de ventilación. Instálese según lo indicado en las instrucciones del fabricante.

No instale el aparato cerca de fuentes de calor tales como radiadores, registros de calefacción, estufas u otros aparatos (incluyendo amplificadores) que produzcan calor.

NO anule la función de seguridad del enchufe polarizado o con clavija de puesta a tierra. Un enchufe polarizado tiene dos patas, una más ancha que la otra. Un enchufe con puesta a tierra tiene dos patas y una tercera clavija con puesta a tierra. La pata más ancha o la tercera clavija se proporciona para su seguridad. Si el toma corriente no es del tipo apropiado para el enchufe, consulte a un electricista para que sustituya el toma corriente de estilo anticuado.

PROTEJA el cable eléctrico para evitar que personas lo pisen o estrujen, particularmente en sus enchufes, en los toma corrientes y en el punto en el cual sale del aparato.

UTILICE únicamente los accesorios especificados por el fabricante.

UTILICESE únicamente con un carro, pedestal, escuadra o mesa del tipo especificado por el fabricante o vendido con el aparato. Si se usa un carro, el mismo debe moverse con sumo cuidado para evitar que se vuelque con el aparato.

DESENCHUFE el aparato durante las tormentas eléctricas, o si no va a ser utilizado por un lapso prolongado.

TODA reparación debe ser llevada a cabo por técnicos calificados. El aparato requiere reparación si ha sufrido cualquier tipo de daño, incluyendo los daños al cordón o enchufe eléctrico, si se derrama líquido sobre el aparato o si caen objetos en su interior, si ha sido expuesto a la lluvia o la humedad, si no funciona de modo normal, o si se ha caído.

NO exponga este aparato a chorros o salpicaduras de líquidos. NO coloque objetos llenos con líquido, tales como floreros, sobre el aparato .



Este símbolo indica que la unidad contiene niveles de voltaje peligrosos que representan un riesgo de choques eléctricos.



Este símbolo indica que la literatura que acompaña a esta unidad contiene instrucciones importantes de funcionamiento y mantenimiento.



Antes de cambiar la alimentación de voltaje o de cambiar el fusible, desconecte el cable de alimentación. Para reducir el riesgo de descargas eléctricas, esta unidad debe ser conectada a tierra.



Remplaze el fusible con lo mismo, que corresponde a lo indicado en el panel del equipo.



Antes de encender, controlar que la línea de alimentación de voltaje corresponda a la indicada.



El interruptor de alimentación es unipolar. En el caso de peligro, desconecte el cable de alimentación. Porque la clavija de conexión a red sirve por la desconexión de la unidad, la clavija debe ser ubicada en proximidad de la unidad.

UNPACKING AND INSPECTION

Your equipment was packed carefully at the factory in a container designed to protect the unit during shipment. Nevertheless, we recommend making a careful inspection of the shipping carton and the contents for any signs of physical damage.

Damage & Claims

If the damage is evident, do not discard the container or packing material. Contact your carrier immediately to file a claim for damages. Customarily, the carrier requires you, the consignee, to make all damage claims. It will be helpful to retain the shipping documents and the waybill number.

Save all packing materials! If You should ever have to ship the unit (e.g. for servicing), it is best to ship it in the original carton with its packing materials because both the carton and packing material have been carefully designed to protect the unit.

Under normal conditions, no user maintenance or calibration is required. Internal links and preset controls may be set to configure the unit during installation. Any service work required should be carried out by qualified service personnel only.

We can offer further product support through our worldwide network of approved dealers and service agents.

To help us provide the most efficient service please would you keep a record of the unit serial number and date and place of purchase to be quoted in any communication regarding this product.

The actual equipment Serial Number is indicated on the silver label stuck on the rear panel of the equipment closure.



Tools and Equipment Needed:

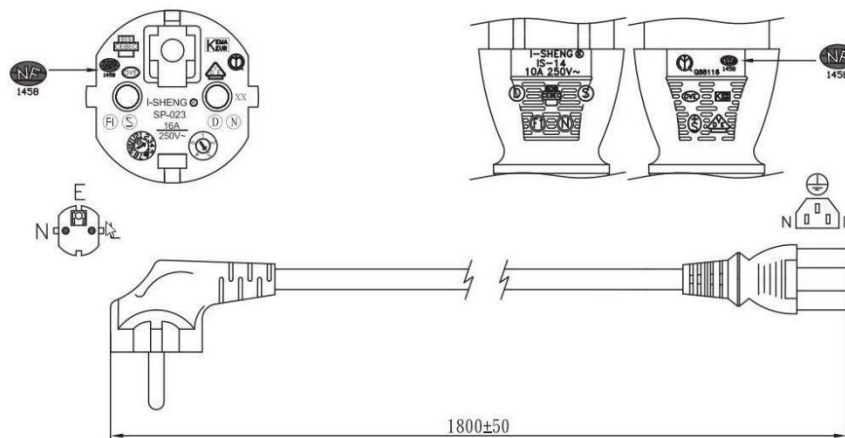
Only standard technician tools are required to install this equipment.

FIRST INSTALLATION RECOMMENDATIONS

POWER SUPPLY

The apparatus is equipped with **2** standard **XLR 4-pin** connectors, namely primary and secondary, to facilitate a redundant power supply setup. It is worth noting that the second external power supply is **OPTIONAL**.

The primary external power supply unit and its cable of approx. **1.8 mt.** lengths are supplied with the device, and the type of the power supply cable plug is a **Hybrid CEE 7/7 plug** (compatible with **E** and **F** socket types).



If for any reason, you need to use this appliance with a different plug, you should use the following wiring guidelines in replacing the existing plug with the new one:

Earth	Green, or green and yellow
Neutral (N)	Blue
Live (L)	Brown

Supply cables should be laid in such a manner that one does not step or walk on them. They should not be squashed by any objects.

POWER SUPPLY

The device is designed for operation with an external power supply of **100 to 240 VAC**, and **50 Hz to 60 Hz**. Check the corresponding device labeling for compatibility with the domestic line voltage and frequency before connecting the IEC power connector to the main supply!



WARNING

Disconnect the mains power plug before you open the housing. Repair of the equipment must only be carried out by authorized and qualified personnel.

THIS EQUIPMENT MUST BE EARTHED.

The chassis is always connected to the mains earth to ensure your safety and the audio quality: check your mains wiring and earthing before switching it on.



PROTECTION AGAINST LIGHTNING



Should the device be put out of action due to being struck by lightning or excess voltage, disconnect it from the power supply without delay. Do not reconnect until the device has been checked. If in doubt contact the technical support service.

Make sure there is suitable lightning protection to protect the device.

Alternatively, you should disconnect all connectors from the device during a storm or when the device is going to be unsupervised or not used for a longer period.

These measures will protect against damage by lightning or excess voltage.

INSTALLATION NOTE AND FIRST STEP

Best setup location

OXYGEN 3000 PLUS should be installed avoiding direct sunlight, close proximity to radiators and air conditioning, dust, water, and chemicals. Choose a console location that permits a clear view of the indicators on the device and ensures sufficient heat dissipation of the device.

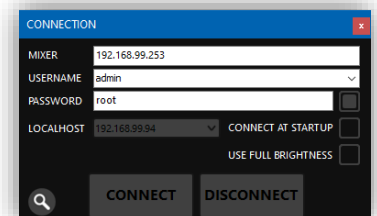
Power Supply	Please make sure that the device and the contained fuse(s) (please see p. 17) are compatible with the domestic line voltage and frequency. If the device is compatible, connect the power supply cord fully to the 4-pin XLR power connector at the rear side of the device and a main power outlet.
Network configuration	OXYGEN has a display, so you can configure the IP settings directly: See step. [MORE INFO] 
Connect to network	Connect a network patch cable to the “10/100-Base-T” connector on the rear side of the device and your existing IP network.
Ready!	These first steps are only intended for a quick first start and do not cover all device functions. Please read carefully the entire manual to be able to use all functions of the device.

FIRST CONNECTION

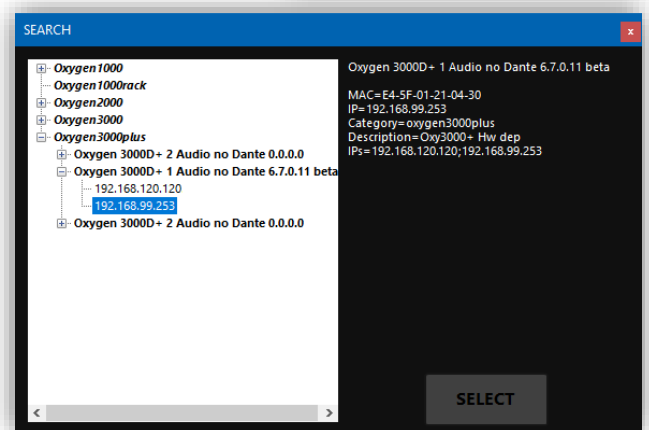
Connect a LAN cable to the **OXYGEN** LAN port on the back panel of the console. The device is automatically discoverable as a **DHCP** client in your network. The **OXYGEN REMOTER** software can be downloaded directly from the dedicated section of the **Axel Technology website**.

Launch the downloaded **OxygenRemoterSetup.exe** installation file. Open Oxygen Remoter by clicking on the **OxygenRemoter.exe** icon. You will see the following window:

Click on the search device button  on the bottom left.



Look for your device in the list and click on it, and click on the **SELECT** button.



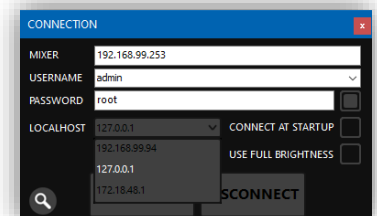
Then click on **CONNECT** button to start **real-time** communication with the target device.

NB:

Default username: **admin**

Default password: **root**

The **Localhost** dropdown menu is used to select from which IP address you need to communicate with the console (in case your PC has more than one IP).



The console is equipped with **two** Internet Protocol (IP) addresses, one of which is assigned as a static IP only (the default IP address is **192.168.120.120**). And the other one comes out pre-set as **DHCP** to facilitate the first connection. Both are fully configurable by navigating to

SETUP / GENERAL / COMMUNICATIONS / TCP-IP

It is possible to **command and monitor** the console using the integrated **display**, **HDMI out**, **Web interface**, and **remoter** software.

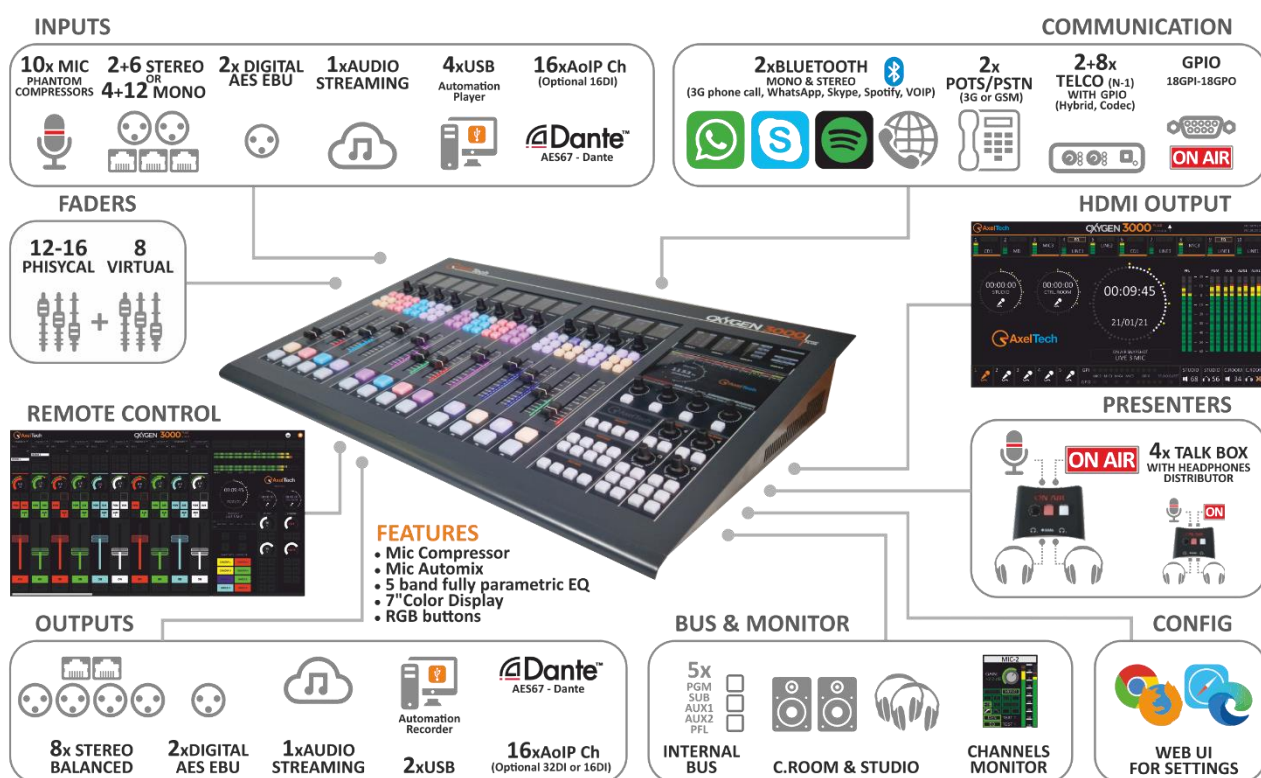


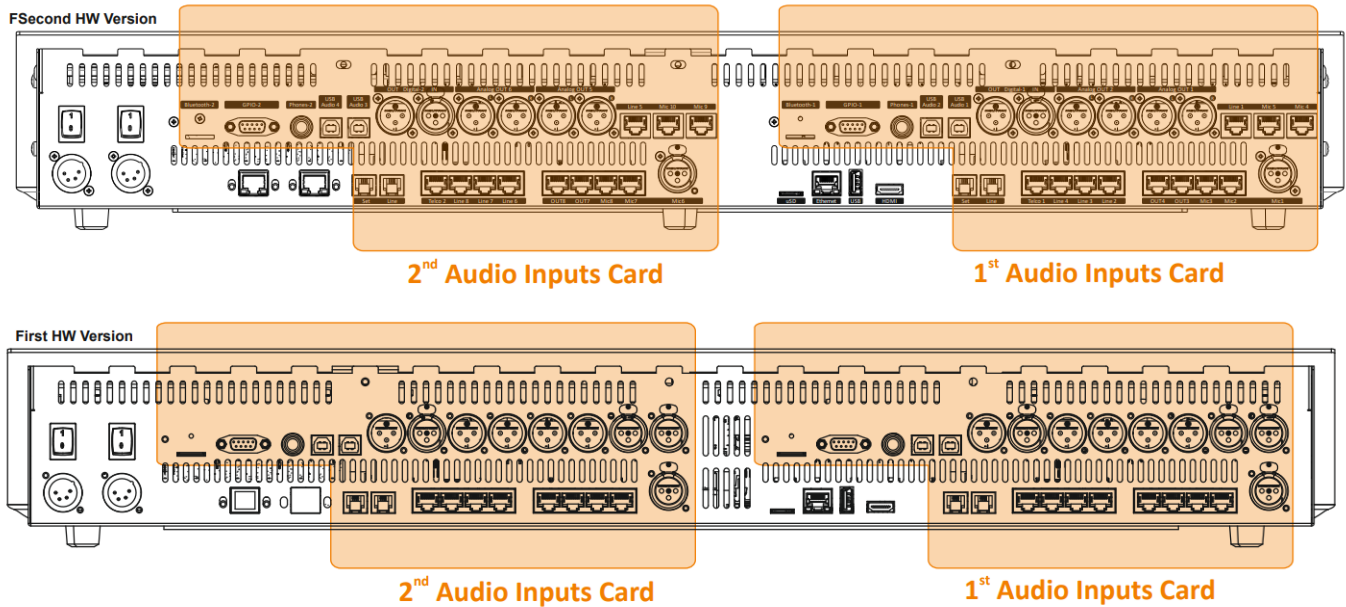
INTRODUCTION

OXYGEN 3000 Plus x1 I/O Board



OXYGEN 3000 Plus x2 I/O Board



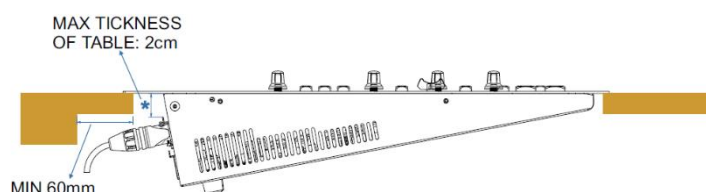
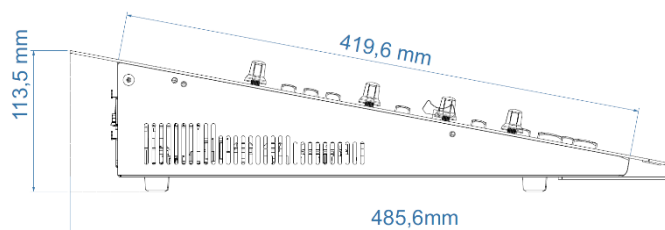
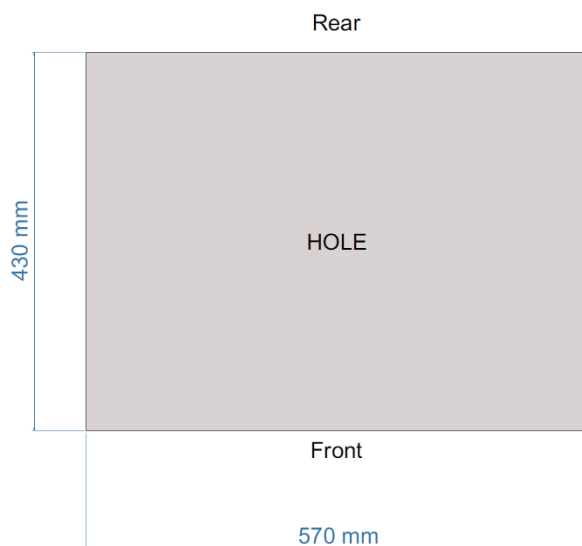
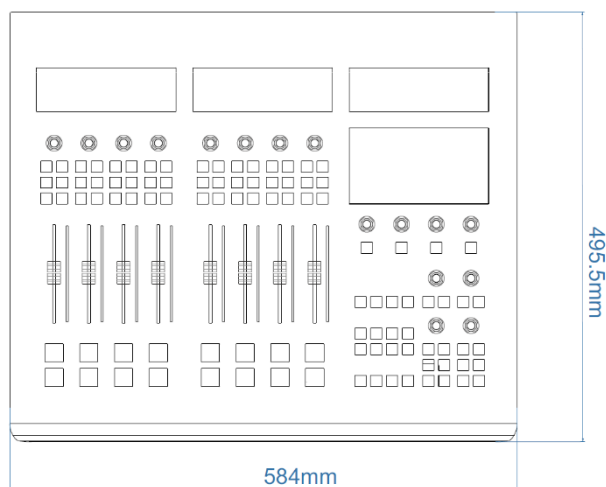


- 8, 12, or 16 faders broadcast digital console
- Analog, digital, USB, AoIP, IP streaming inputs and outputs
- Digital audio routing
- AoIP interface Dante/AES67 (optional)
- Audio Streaming Encoder/Decoder
- User definable Smart Keys for shortcuts and macro commands
- Software for remote control and configuration/settings
- Web interface for configuration/settings
- Backlight RGB buttons
- 16 displays 2" 240x 320 IPS showing channel status, levels, and settings
- 1 display 7" TFT IPS for settings and configuration
- HDMI output
- Latency < 1mSec
- Start-up time <10sec.
- Backup SD Card
- Switching PSU 90/260 Vac 50/60Hz (redundant option available)

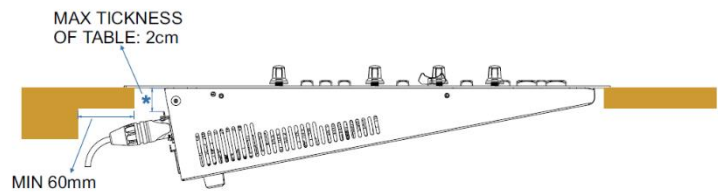
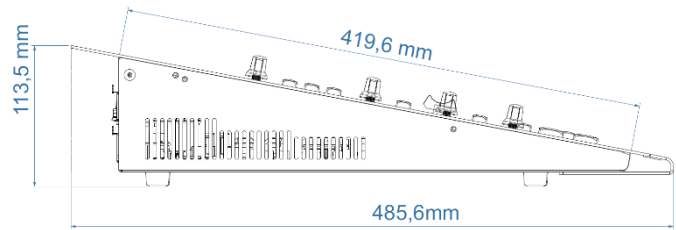
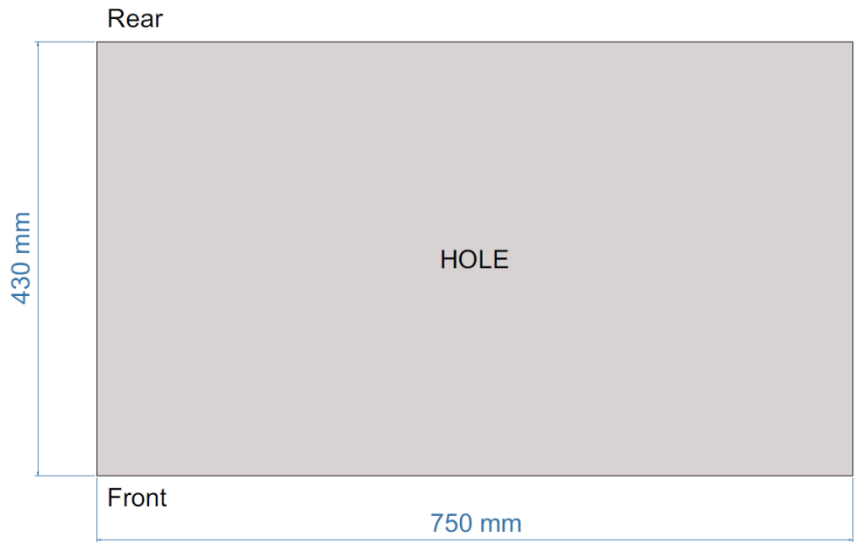
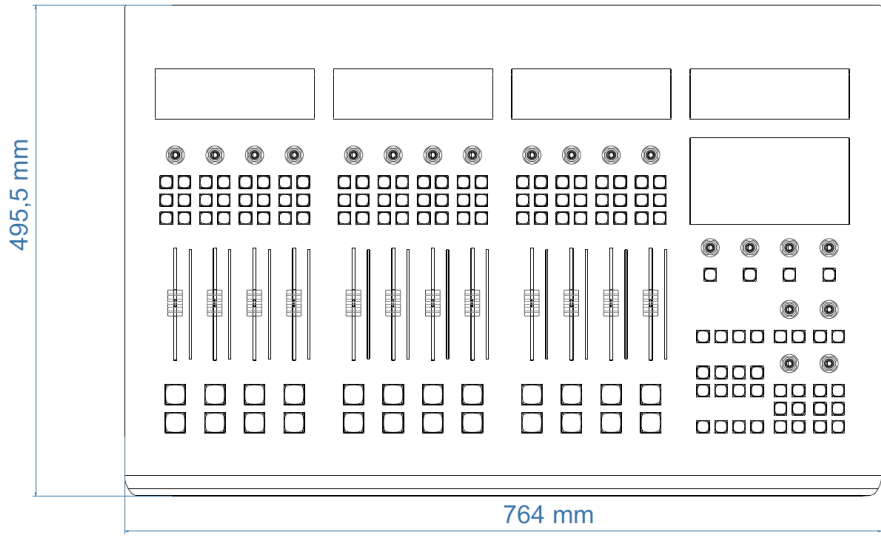
The 2° Audio Input Card is **Optional**.

GENERAL DESCRIPTION

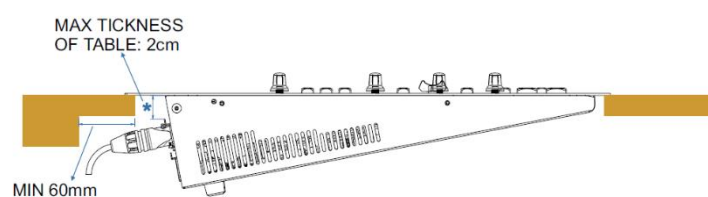
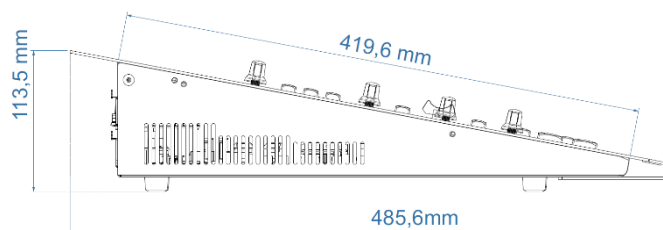
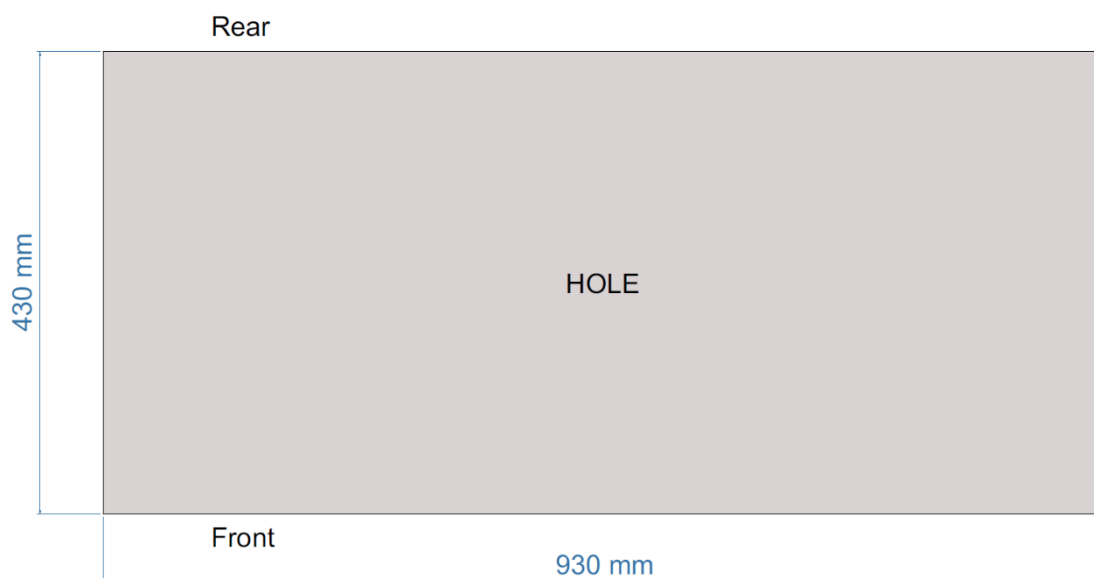
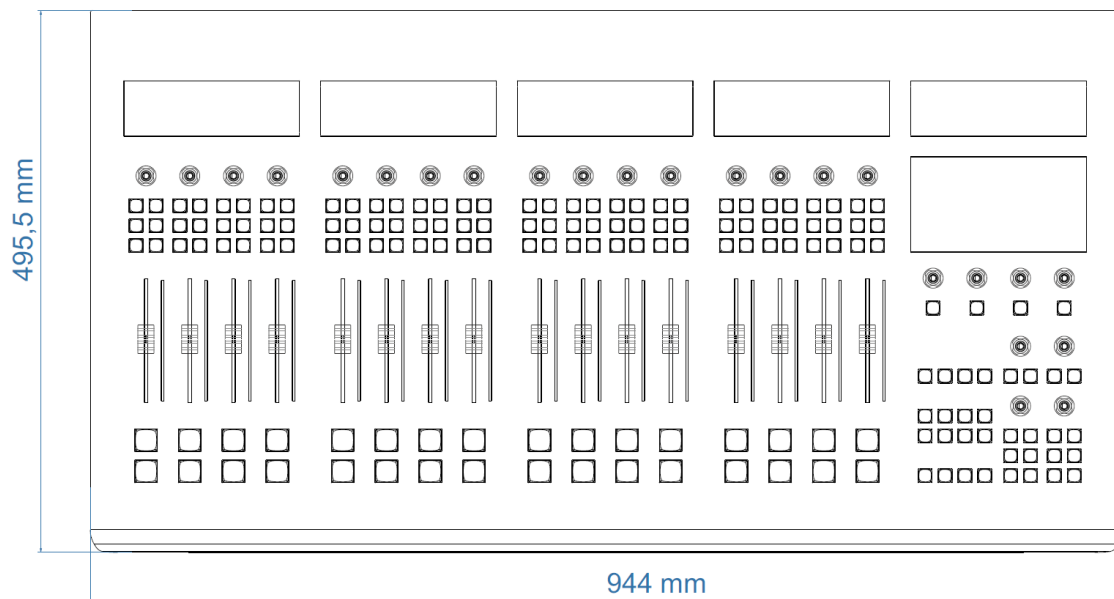
DIMENSIONS (8 PHYSICAL FADERS)



DIMENSIONS (12 PHYSICAL FADERS)

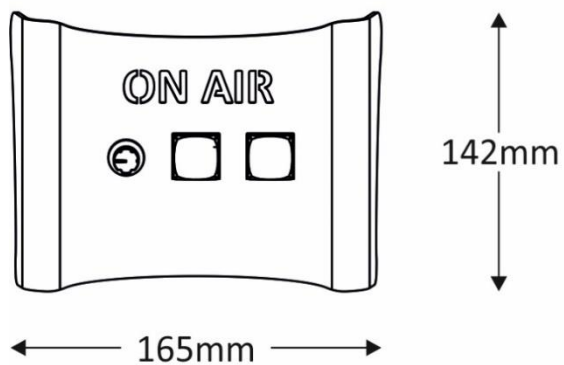


DIMENSIONS (16 PHYSICAL FADERS)

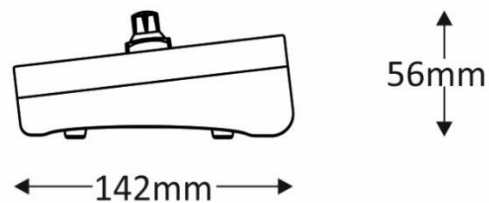


TALKBOX DIMENSIONS

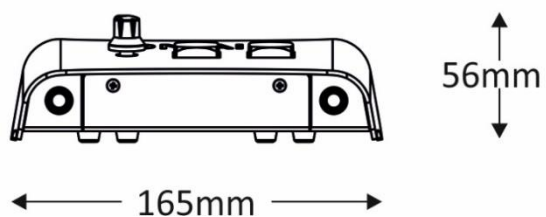
TOP



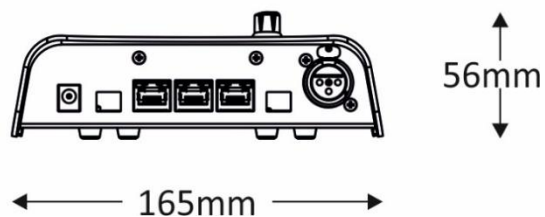
SIDE



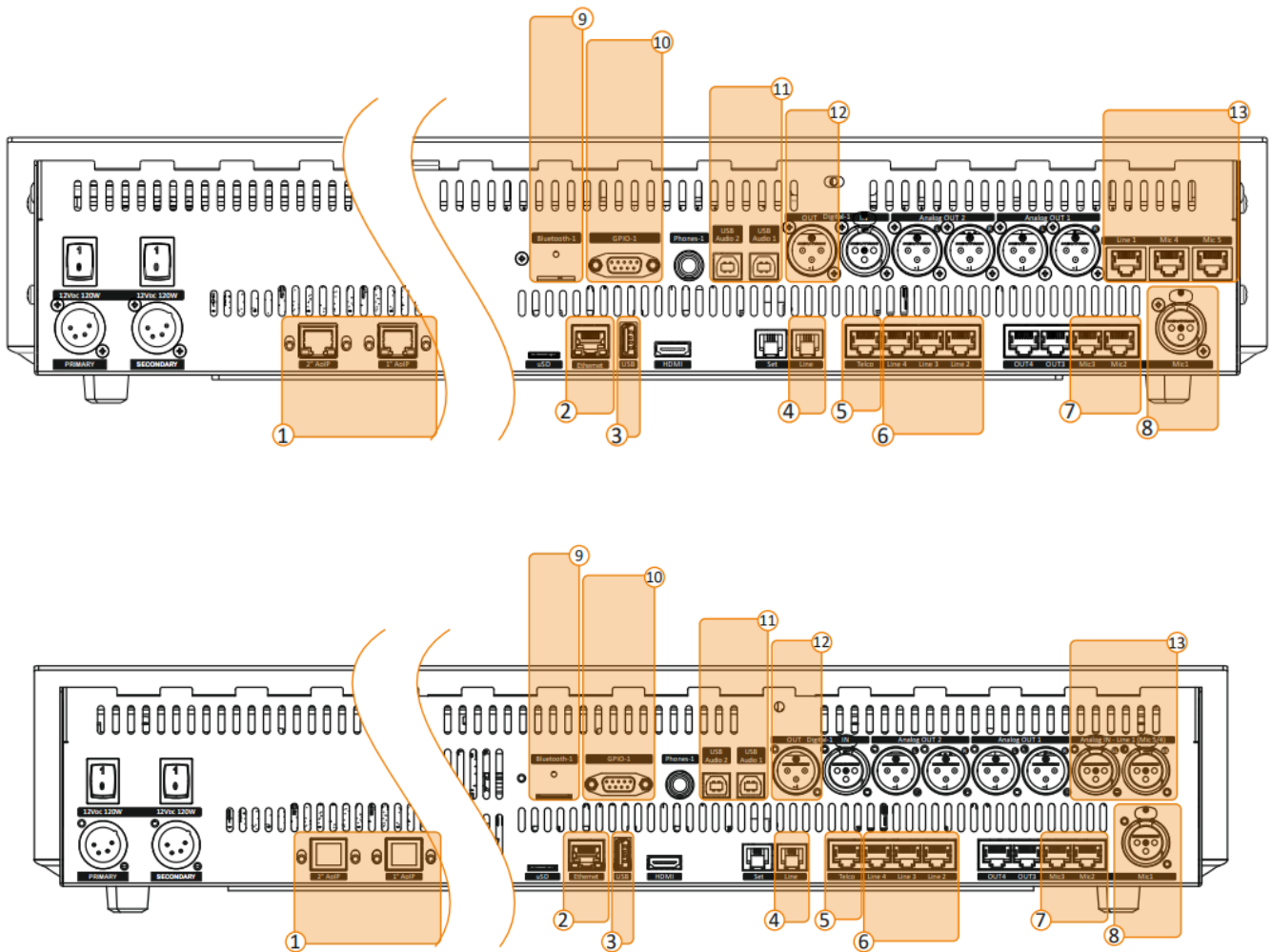
FRONT



REAR



INPUT CONNECTIONS



1. AOIP – LAN DANTE (OPTIONAL).

16 or 32 Stereo digital audio inputs over Ethernet. If the DANTE board was not purchased with the device these inputs are not enabled and do not work.

Each DANTE channel could be set as:

Stereo, 2 mono, or 2 Telco Inputs. updates.

[MAIN / AUDIO / SETTINGS / INPUT MODE / DANTE / INPUT / MODE](#)

2. Ethernet

RJ45 Female Connector Internet Connection used as **streaming audio input**.

Used also for controlling the console via the Web interface, remote control, and online updates.

[MAIN / GENERAL SET. / COMMUNICATIONS / TCP-IP](#)

3. USB

A USB port Type A to **Export** and **Import** the mixer configurations and to customize the station LOGO. To configure it navigates to [MAIN / AUDIO / SETTINGS / INPUT MODE](#)

7. MIC 2, MIC 3

MIC input over the RJ45 connector includes **2 GPI** and **1 GPO** on each connector. These 2 ports can be used to connect the **Talk Box** interfacing or connect the mic directly by a special cable called **EASY MIC**.
Balanced Audio Connection (10 kΩ).

8. MIC 1

Mic Input - XLR Female - Balanced Audio Connection (1.2kΩ).

9. BT

BT Stereo/Mono Input.

Wireless Smartphone **input/output** connection.

To configure, it navigates to [MAIN / AUDIO / INPUTS / BT](#)

10. GPIO

SUB-D 9p Female, **4 GPI** & **4 GPO**.

To configure it navigates to [MAIN / GENERAL SET. / GPIO](#)

4. Telephone

Analog Telephone Line Input (RJ11) for **POTS/PSTN** interfacing. To configure it navigates to [MAIN / AUDIO / INPUTS / PHONE](#)

5. Telco I/O

TELCO Input/output over RJ45 connector. It is also carried **2 GPIO** signals for controlling external telephone hybrid or sending status information.

1 GPI to get the incoming call signal by the flashing of the F1 button.

1 GPO to control the external hybrid device for the hook or drop of the calls.

To configure it navigates to [MAIN / AUDIO / INPUTS / TELCO](#)

6. ANALOG IN (LINE-2, LINE-3, LINE-4)

3 Stereo analog audio inputs over RJ45 connectors. Each one of these inputs is configurable to be as: **STEREO** by default, **2 MONO**, or **2 TELCO**. The user can change the mode of the source by navigating to [MAIN / AUDIO / SETTINGS / INPUT MODE](#)

11. USB AUDIO 1, USB AUDIO 2

2 Audio Card Stereo Input/output **bidirectional** Type **USB-B**.

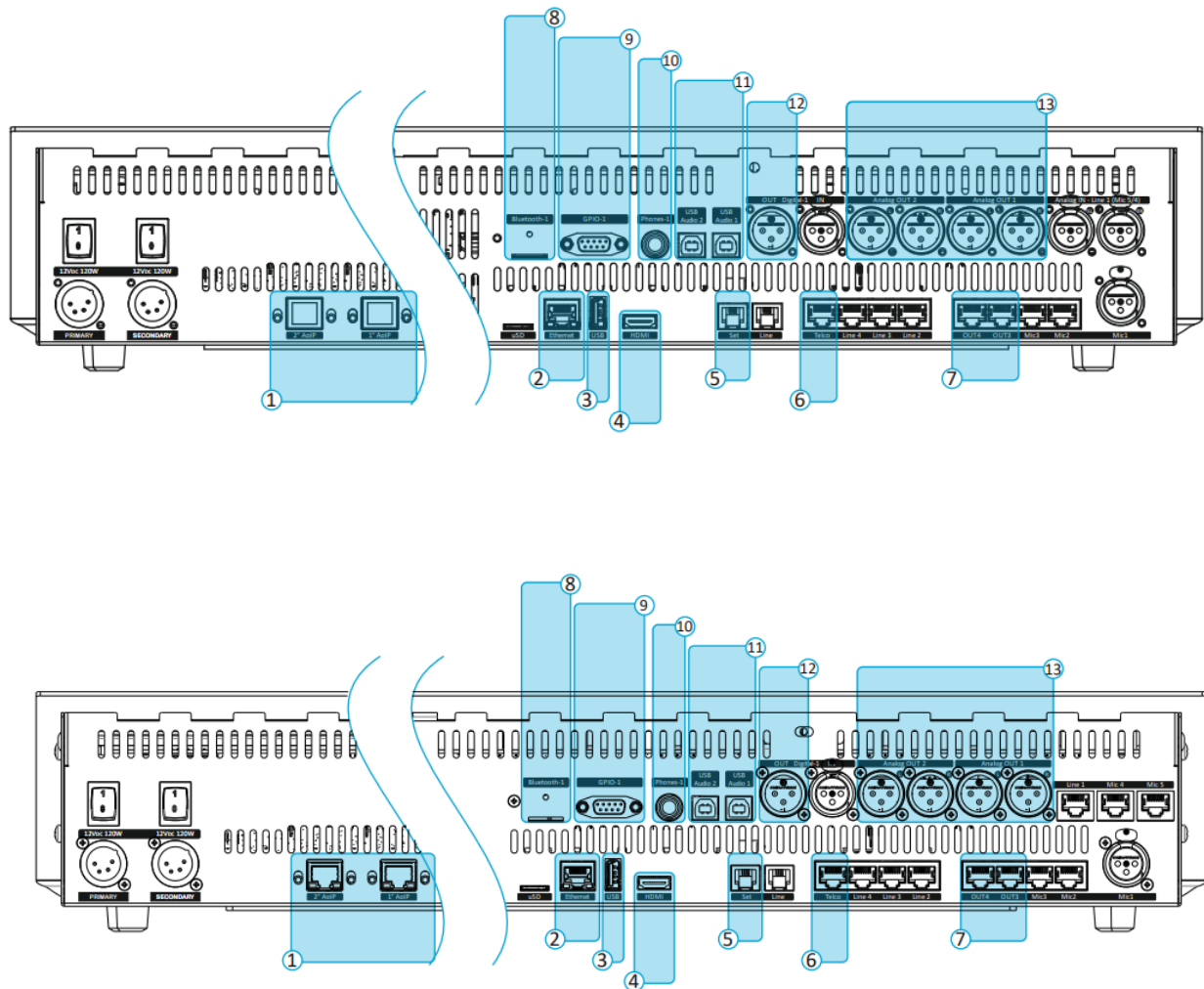
12. DIGITAL INPUT

(AESEBU) on XLR Female connector for the digital AESEBU input audio signal. Balanced Audio Connection (110Ω)

13. LINE 1 / MIC 4, MIC 5

This audio input section includes three connectors. The **left** and **centre connectors** can be used for **MIC** or **MONO** signals, while the **right-hand connector** is dedicated to **STEREO** signals.

OUTPUT CONNECTIONS



1. **AOIP – LAN DANTE (OPTIONAL).**
16 or 32 Stereo digital audio outputs over Ethernet. If the DANTE board was not purchased with the device these outputs are not enabled and do not work.
Each DANTE channel could be set as:
Stereo, 2 mono, or 2 Telco outputs. updates.
[MAIN / AUDIO / SETTINGS / INPUT MODE / DANTE / INPUT / MODE](#) [\[MORE INFO\]](#)
2. **Ethernet**
RJ45 Female Connector Internet Connection used as **streaming audio output**.
[MAIN / AUDIO / SETTINGS / INPUT MODE / STREAMING](#) [\[MORE INFO\]](#)
3. **USB**
A USB port Type A to **Export** and **Import** the mixer configurations. To export the configuration navigate to [MAIN / AUDIO / OUTPUTS / DIGITAL / USB](#) [\[MORE INFO\]](#)
4. **HDMI Output**
5. **Bluetooth**
6. **GPIO**
7. **ANALOG OUTPUT (OUT 3, OUT 4)**
2 Stereo analog audio outputs over RJ45 connectors. Each one of these outputs is configurable to be as: **PGM, SUB, AUX, SPEAKER, HEADPHONE, or TELCO(n-1)**. The user can change the mode of the source by navigating to [MAIN / AUDIO / OUTPUTS / ANALOG](#) [\[MORE INFO\]](#)
8. **BT**
Wireless Smartphone **input/output** connection. BT Mono phone call communications, Skype, WhatsApp, ...etc.
To configure, it navigates to [MAIN / AUDIO / INPUTS / BT](#) [\[MORE INFO\]](#)
9. **GPIO**
SUB-D 9p Female, **4 GPI & 4 GPO**.
To configure it navigates to [MAIN / GENERAL SET. / GPIO](#) [\[MORE INFO\]](#)
10. **Phone**
11. **Audio**
12. **Audio**
13. **Audio**

Standard HDMI Female Connector – External Monitor.
To configure it navigates to [MAIN / GENERAL SET. / LIGHT&DISPLAY](#) [\[MORE INFO\]](#)

5. Telset

(RJ11) used to connect the Telset port to a telephone base unit.

6. Telco I/O

TELCO Input/output over RJ45 connector. It is also carried **2 GPIO** signals for controlling external telephone hybrid or sending status information.

1 GPI to get the incoming call signal by the flashing of the F1 button.

1 GPO to control the external hybrid device for the hook or drop of the calls.

To configure it navigates to [MAIN / AUDIO / INPUTS / TELCO](#) [\[MORE INFO\]](#)

10. HEADPHONES

On a female Jack 6,3 mm connector for the audio monitoring with Control Room Headphones.

11. USB AUDIO 1, USB AUDIO 2

2 Audio Card Stereo Input/output **bidirectional** Type **USB-B**.

12. DIGITAL OUTPUT

(AESEBU) on XLR Male connector for the digital AESEBU output audio signal. Balanced Audio Connection (110Ω)

13. ANALOG-OUT-1, ANALOG-OUT-2

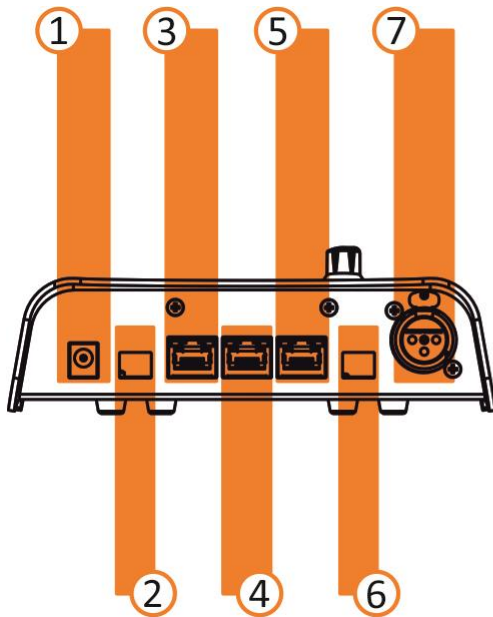
2 Stereo Output - XLR Male - Balanced Audio Connection (47Ω)

Each one of these outputs is configurable to be as:

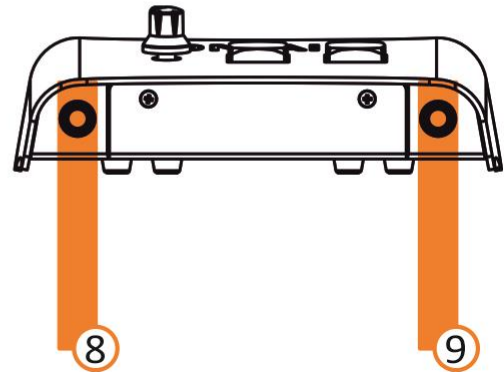
PGM, SUB, AUX, SPEAKER, HEADPHONE, or TELCO(n-1).

The user can change the mode of the source by navigating to [MAIN / AUDIO / OUTPUTS / ANALOG](#) [\[MORE INFO\]](#)

TALKBOX CONNECTIONS

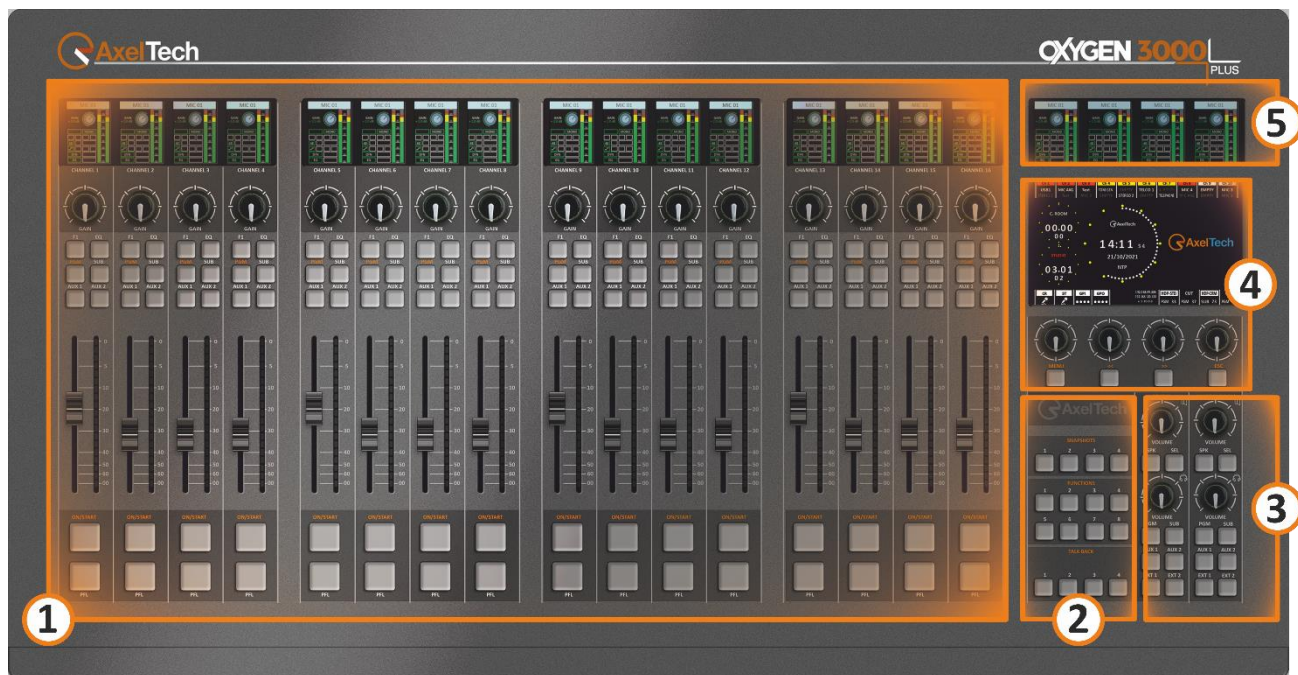


1. **Power Supply**
12VDC 1A
2. **ON AIR Light relay**
DC relay normally opens NO.
2 PIN Screw Connector.
3. **HDP Output**
RJ45 Connector (**SFTP CABLE**
REQUIRES) - (Passive Loop Output).
4. **HDP Input**
RJ45 Connector (**SFTP CABLE**
REQUIRES)
5. **Talk Box Connection**
MIC & GPIO connection over RJ45
Connector (**SFTP CABLE** REQUIRES).
(Passive microphone Output).



6. **Mic Light**
2 PIN Screw Connector - (12VDC
output).
7. **Mic Input**
XLR Female Connector.
8. **HDP-1-TBox**
Jack 6.3mm Female Connector -
(Min. Imp. 32Ω).
9. **HDP-2-TBox**
Jack 6.3mm Female Connector -
(Min. Imp. 32Ω).

SURFACE



1. CHANNEL CONTROLS

Depending on the selected configuration, the console provides **8, 12 or 16 physical channel strips**, with up to **8 additional virtual faders** available through OXYGEN REMOTER. Each physical channel strip incorporates a **2.2" TFT IPS display**, which shows the selected source, channel status, configuration information and **pre-fader audio level**. The associated controls provide direct access to source selection, gain, bus assignment, channel level, PFL monitoring and ON/OFF operation.

2. SPECIAL FUNCTION BUTTONS

The **SPECIAL FUNCTION BUTTONS** provide direct access to frequently used operational and communication functions. This section includes **4 SNAPSHOT buttons**, **8 configurable SMART KEYS** and **4 TALKBACK buttons**. The SNAPSHOT buttons recall assigned console configurations, while the SMART KEYS can perform user-defined functions or transmit **TCP, UDP or REST API commands** to compatible external devices and software. The TALKBACK buttons activate the configured communication paths between the Control Room, Studio and other assigned destinations.

3. MONITORS SECTION

The **MONITORS SECTION** manages the loudspeaker and headphone-monitoring paths for both the **CONTROL ROOM** and the **STUDIO**. Independent rotary controls adjust the listening levels, while pressing the corresponding encoder mutes or unmutes the selected monitoring output. The source-selection buttons provide direct monitoring of **PGM, SUB, AUX 1, AUX 2, EXT 1 and EXT 2**. Depending on the configured monitor mode, the **PFL** function can temporarily make the pre-fader signal of one or more channels audible through the selected loudspeakers or headphones.

4. MENU CONTROL — BUTTONS & DISPLAY 7" TFT:

The **MENU CONTROL** section consists of a **7" TFT IPS colour display**, four rotary/push encoders and the dedicated **MENU**, **>>**, **<<** and **ESC** navigation buttons. The four encoders correspond to the parameters shown on the display. Rotating an encoder changes the associated value, while pressing it confirms and stores the selection. The display provides access to console configuration menus and shows operating information, channel states, source names, monitoring status, clock, microphone timers, firmware release and network information. An **HDMI output** is also available for displaying console status on an external screen.

5. OUTPUTS DISPLAY:

The **OUTPUTS** area incorporates **four 2.2" TFT IPS displays** used to provide continuous visual monitoring of the console's principal output buses. The displays show stereo audio levels and operating information for **PGM**, **SUB**, **AUX 1**, **AUX 2** and **PFL**. They also provide status information for the Control Room and Studio loudspeakers and headphones. Depending on the active display layout, additional information such as date, time and microphone timers may also be shown.

CHANNEL CONTROLS

CHANNEL CONTROLS identifies the physical control strip associated with each mixer channel. It is the main area from which the operator manages the selected audio source during normal console operation.

Through this section, it is possible to:

- Display the selected source and its audio level
- Adjust the input gain and channel volume
- Select or change the active source
- Send the signal to the required output buses
- Activate or deactivate the channel
- Listen to the signal using the PFL function
- Access additional processing and communication functions

The **CHANNEL CONTROLS** section consists of the following elements.

Meaning of illuminated buttons

The illuminated buttons provide an immediate indication of the status of the associated function. Their exact colour and behaviour may depend on the console configuration and firmware, but the general meaning is as follows:

- **Button off:** the associated function is normally inactive, disabled or not selected.
- **Button steadily illuminated:** the associated function is active, selected or enabled.
- **Button flashing:** the function may be in a temporary, pending, warning or attention-required state. Depending on the control, flashing may indicate a source starting, a call or communication request, a status change, a PFL condition, or another function requiring operator attention.



CHANNEL DISPLAY:

The **CHANNEL DISPLAY** is the display associated with the channel. It provides the operator with the main information concerning the source and the channel's operating status. It may display:

- Name or identification of the selected source
- Active source between **CH-A** and **CH-B**
- Channel status, such as Standby or On Air
- Pre-fader audio level
- Status of the assigned output buses
- Active or available functions
- Information concerning the channel configuration

The display allows the operator to quickly check the channel configuration without accessing the console's general menus. In this way, the source, level and operating status can be monitored directly from the control surface. The displayed level normally refers to the signal before the fader. Therefore, the indicated value may not correspond to the actual level sent to the output after fader adjustment.

GAIN/KNOB:

The **GAIN** control adjusts the level of the signal coming from the selected source. Its purpose is to adapt the input signal to the correct operating range of the console.

A correct **GAIN** setting allows the operator to:

- Obtain a sufficient level for subsequent audio processing
- Avoid noise caused by an input level that is too low
- Avoid distortion caused by an excessive level
- Keep the signal within an appropriate operating margin
- Ensure more effective adjustment through the fader

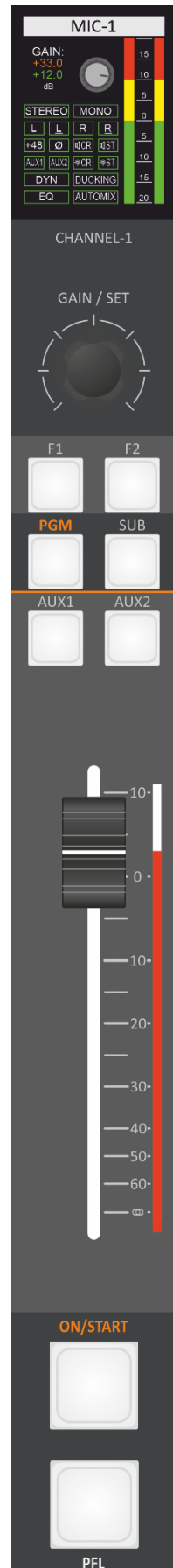
GAIN should not be used as the normal volume control during transmission. The gain should be set during system configuration or source checking, while the daily operating level should mainly be managed using the fader. The **GAIN** control also allows the operator to select or switch between:

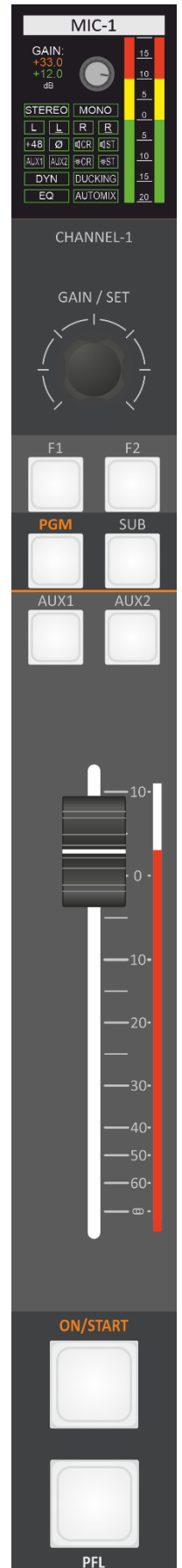
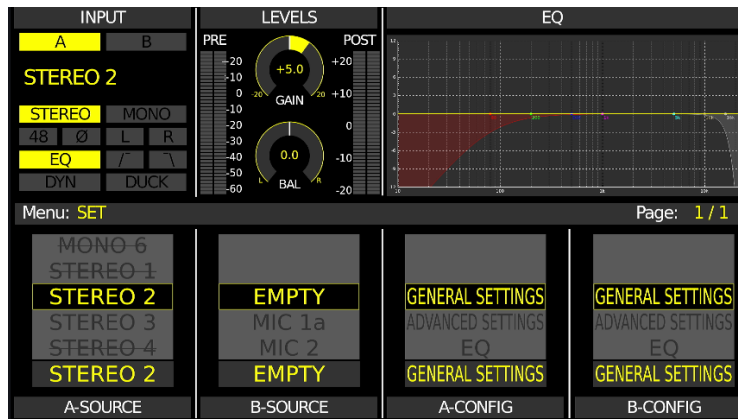
- **CH-A**, the primary source
- **CH-B**, the secondary source

The availability and behaviour of CH-A and CH-B depend on the channel configuration and the type of connected source.

ATTENTION: To quickly configure a **CHANNEL**, press and hold the corresponding **GAIN/KNOB** for the required **CHANNEL 1–8** for a few seconds.

Warning: excessive gain can cause clipping and distortion before the signal reaches the fader. Insufficient gain can increase noise and reduce signal quality.





F1 and F2:

The **F1** and **F2** buttons are configurable controls associated with the individual channel. Their function may vary according to the selected source, the console configuration and the operating mode.

They may be used, for example, to recall or activate:

- Equalization
- Compressor
- Ducking
- Talkback
- Telephone-line functions
- Communication functions
- Other processes or commands assigned by the user

The actual function of each button must be checked on the display or in the channel configuration, since F1 and F2 do not necessarily have a fixed function on every channel. These buttons provide quick access to functions used during programme operation or technical management, avoiding the need to open the configuration menus each time.

Note: the function of F1 and F2 may differ from one channel to another. Before using them during a live programme, verify the button assignments.

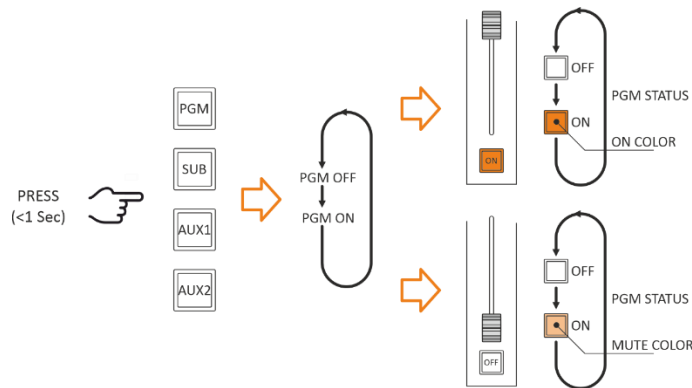
PGM, SUB, AUX1 and AUX2:

The **PGM**, **SUB**, **AUX1** and **AUX2** controls determine the routing of the channel signal to the console's output buses. The main assignments are:

- **PGM:** sends the signal to the main programme bus
- **SUB:** sends the signal to the secondary or sub-programme bus
- **AUX1:** sends the signal to auxiliary bus 1
- **AUX2:** sends the signal to auxiliary bus 2

These controls determine where the signal is sent within the console. Bus assignments can be used to create different operating combinations, such as:

- Main programme
- Sub-programme
- Monitoring
- Recording
- Sending to external systems
- Communication or technical contributions
- Feeding additional devices or audio paths



Assigning a channel to a bus does not necessarily activate the channel. A channel may be assigned to one or more buses, but the signal is actually sent according to the channel status, the fader position and the internal console configuration. The combination of buses allows the main path and auxiliary paths to be managed separately, providing greater flexibility during production and transmission.

FADER:

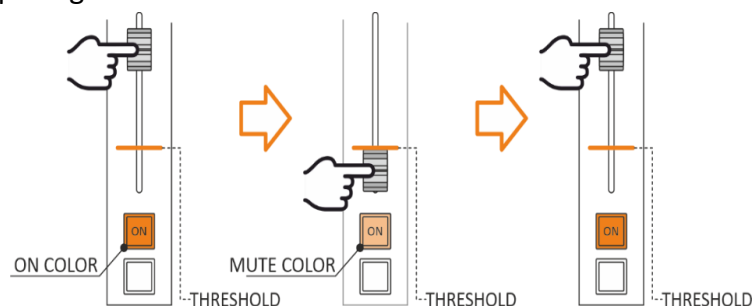
The **FADER** adjusts the operating level of the channel. It is the control normally used by the operator to increase or decrease the source volume during transmission. The fader affects the level of the signal sent to the assigned buses, according to the channel configuration and its audio path. It can be used to:

- Gradually bring a source On Air
- Increase or reduce the level of a voice or music source
- Perform fades
- Balance several sources simultaneously
- Temporarily remove a source by lowering its level
- Adjust the relationship between speech, music, contributions and other sources

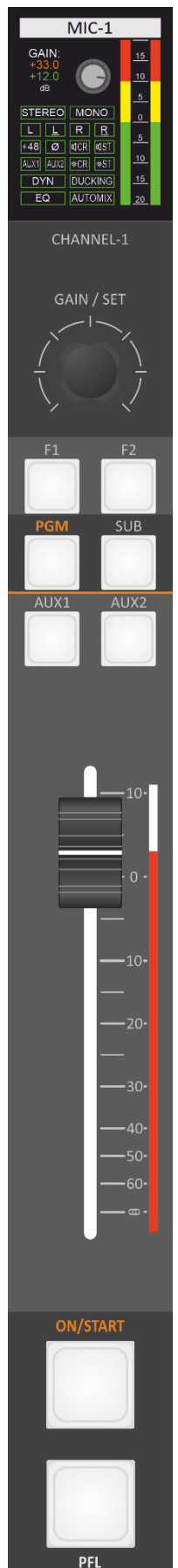
The fader must be distinguished from GAIN:

- **GAIN** adapts the level of the input signal
- **FADER** adjusts the channel's operating level during use

A correct GAIN setting allows the fader to be used within a suitable operating range, maintaining sufficient adjustment reserve. The fader position may also be used by the console to determine operating conditions, such as activating On Air logic, starting sources or controlling functions associated with the channel, according to the installed configuration.



FADER LED BAR:



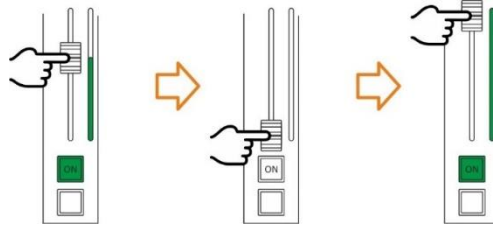
The **FADER LED BAR** is the illuminated bar associated with the fader. It provides an immediate visual reference of the control position and channel status. It may indicate:

- Fader position or attenuation
- Channel operating status
- Source type
- On Air or Standby condition
- Functions assigned through configurable colours

The LED bar allows the operator to quickly identify the channel status without reading the display. It is particularly useful when several channels must be monitored simultaneously.

Colours may be used to distinguish source categories, such as:

- Microphones
- Audio players
- Telephone lines
- External connections
- Service sources
- Virtual or remote channels



The meaning of the colours depends on the console configuration. For this reason, the colour coding should be defined consistently and documented for all operators.

ON/START:

The **ON/START** control manages channel activation and the transition from Standby to the operating or On Air condition. Depending on its illumination status, the **ON/START** button performs different functions:

- 1) **LED OFF** - OFF status - the channel is MUTE.
- 2) **LED ON** in (warm color) - ON status – the channel is OPEN.
- 3) **LED ON** in (light color) – standby status.



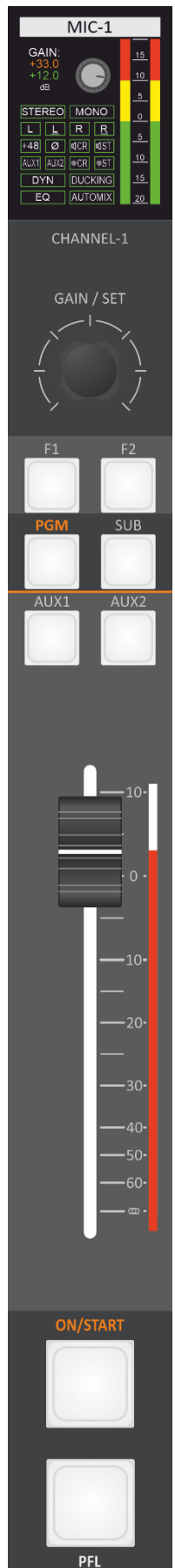
When the channel is activated:

- The source may be sent to the assigned buses
- The channel may become active in the transmission
- Functions associated with the source may be activated
- The corresponding status may be shown on the display
- Automation or control logic may be activated

The exact behaviour of the control depends on the source type and channel configuration. In some cases, ON/START may also start a source, player or function connected to the channel.

When the channel is deactivated:

- The source is removed from the operating path according to the configuration
- The channel returns to Standby
- Associated functions may be deactivated
- The signal may no longer be sent to the output buses



Before activating a channel, check the signal level, bus assignments and the possible presence of open microphones or unintended sources.

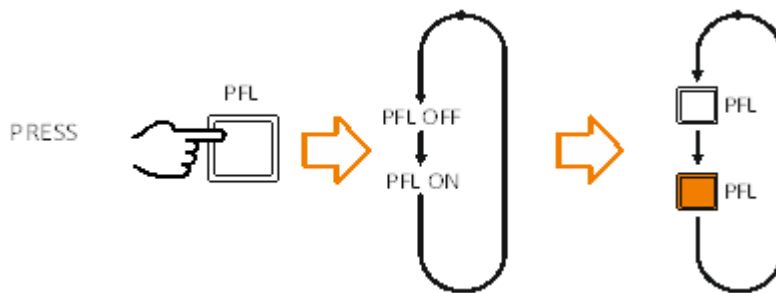
PFL:

The **PFL**, or **Pre-Fader Listen**, control allows the operator to listen to the channel signal before the fader.

The PFL function is used to:

- Check a source without changing its On Air level
- Verify the presence of the signal
- Correctly adjust GAIN
- Identify distortion, noise or connection problems
- Prepare a source before bringing it On Air
- Check a channel that is not yet active

PFL monitoring is independent of the fader position. Therefore, the operator can check the signal even when the fader is lowered or the channel is not yet On Air.

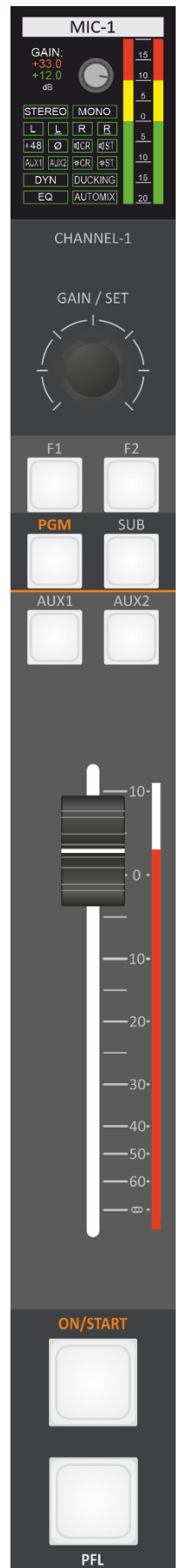


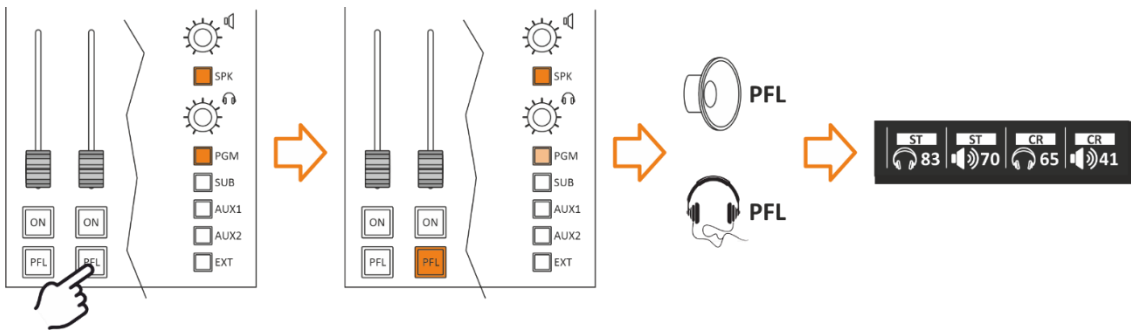
The console may provide two different PFL operating modes:

- **SINGLE PFL:** only one selected channel is monitored at a time. Activating PFL on another channel normally replaces the previously selected PFL signal.
- **SUM PFL:** the PFL signals of several selected channels are combined and monitored simultaneously. This allows the operator to listen to more than one source at the same time, while each selected channel contributes to the common PFL monitoring signal.

The difference is therefore that **SINGLE PFL** provides exclusive monitoring of one channel, whereas **SUM PFL** creates a combined monitoring signal from multiple channels.

SINGLE PFL is useful when the operator needs to isolate and accurately check one source, for example during gain adjustment or fault diagnosis. **SUM PFL** is useful when several sources must be monitored together, for example when checking the balance or presence of multiple channels before or during a programme.





For telephone sources, PFL may also allow private off-air communication with the caller. In this case, the operator can speak to the caller without sending the conversation to the main programme.

Warning: activating PFL changes the technician's monitoring path. Before starting transmission, verify that PFL has been deactivated when it is no longer required. When using SUM PFL, also verify which channels are selected, since more than one source may be present in the monitoring signal.

PRIVATE MIC:

if you press the **PFL** (Pre-Fader Listen) button on the telephone channel, any microphone that is set as a private microphone will be able to communicate with the caller in private, **even if the fader is closed**. This can be useful in situations where you need to have a private conversation with the caller without it being heard ON-AIR audio feed.

Overall operation of CHANNEL CONTROLS:

The CHANNEL CONTROLS operate together according to the following typical sequence:

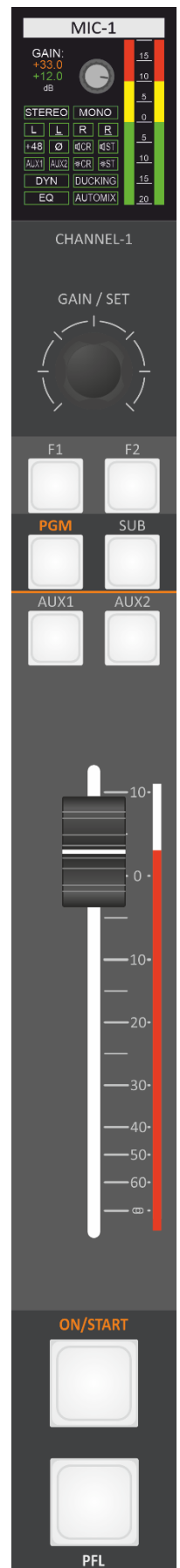
1. Select the source using CH-A or CH-B.
2. Adjust GAIN to obtain the correct input level.
3. Check the signal using CHANNEL DISPLAY and PFL.
4. Assign the channel to the PGM, SUB, AUX1 and AUX2 buses.
5. Adjust the level using the FADER.
6. Activate the channel using ON/START.
7. Monitor the channel visually using the FADER LED BAR and CHANNEL DISPLAY.
8. Use F1 and F2 for the configured additional functions.

CHANNEL CONTROLS therefore combines, in a single physical strip, all the functions required for the daily management of a source: selection, adjustment, routing, monitoring, activation and visual control.

Number of channels:

Depending on the console model, OXYGEN 3000 Plus may provide:

- 8 physical channel strips
- 12 physical channel strips
- 16 physical channel strips
- Up to 8 additional virtual channels through OXYGEN REMOTER



Virtual channels can be managed through the remote interface and increase the number of controllable sources without adding further physical strips to the console surface. The actual configuration of the channels, sources and available functions depends on the installed model, firmware and system settings.

SPECIAL FUNCTION BUTTONS

SPECIAL FUNCTION BUTTONS identifies the physical control area used to perform global operations that are not associated exclusively with an individual mixer channel. This section allows the operator to:

- Recall previously saved channel configurations
- Change several console settings through a single command
- Control compatible external software or equipment
- Send TCP, UDP or REST API commands
- Activate Talkback communications
- Communicate with presenters, guests and technical personnel
- Quickly access functions required during normal production

The SPECIAL FUNCTION BUTTONS section consists of the following elements.

SNAPSHOTS:

The **SNAPSHOTS** section consists of four physical buttons used to recall previously saved channel configurations. Each button can be associated with a specific snapshot stored in the console. Pressing the corresponding button recalls the assigned configuration and applies the parameters included in that snapshot. The console can store up to **10 channel snapshots**.

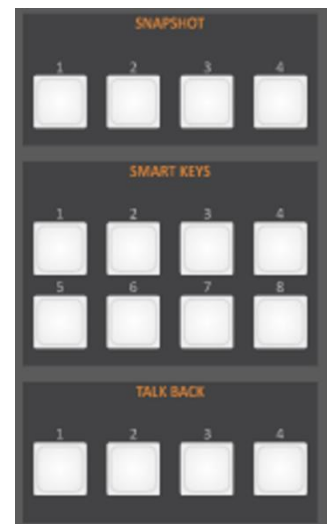
Each snapshot can contain settings such as:

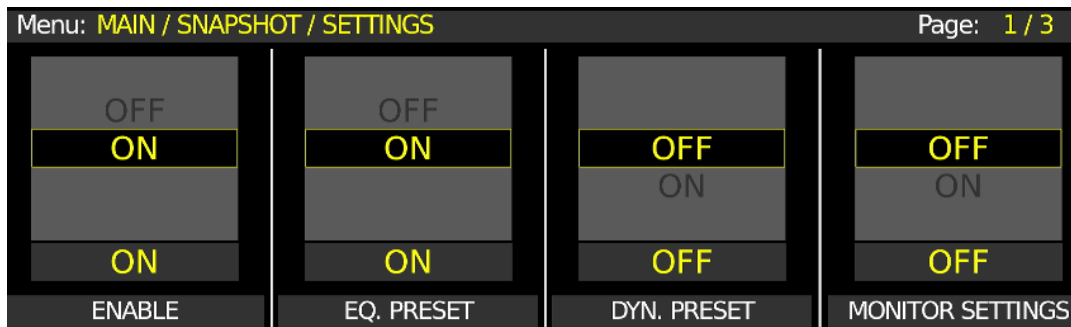
- CH-A and CH-B source assignments
- Equalizer settings
- Compressor and dynamics settings
- PGM, SUB, AUX1 and AUX2 bus assignments
- Fader levels
- Source gain levels
- Pre-microphone levels
- Channel ON/OFF states
- Channel F1 and F2 assignments
- Monitoring and output settings, depending on configuration

The parameters restored by a snapshot depend on the options enabled in the snapshot's advanced settings.

Snapshot configuration can normally be accessed through: **MAIN → SNAPSHOT**

The advanced parameter selection can be accessed through: **MAIN → SNAPSHOT → ADV. SETTINGS**





The snapshot menu allows the technician to:

- Select one of the available snapshot positions
- Assign a recognizable name
- Save the current channel configuration
- Recall a stored snapshot
- Associate a snapshot with a physical recall button
- Select which parameter groups must be restored
- Delete a snapshot that is no longer required

The four physical buttons provide direct access to four of the stored snapshots. The remaining snapshots can still be recalled through the console menu. SNAPSHOTS may be used to prepare different operating configurations, such as:

- Normal On Air operation
- News programme
- Interview or talk-show configuration
- Music programme
- Podcast production
- Recording session
- Outside broadcast
- Emergency or backup configuration

For example, one button may recall the normal On Air configuration, while another button recalls a configuration prepared for an interview with several microphones.

Warning: recalling a snapshot may change source assignments, gains, fader levels, bus assignments and channel states. Before pressing a SNAPSHOT button during a live programme, verify which parameters are included and which preset is assigned to the button.

SMART KEYS / FUNCTIONS:

The **SMART KEYS** section consists of eight configurable buttons.

Their standard names are:

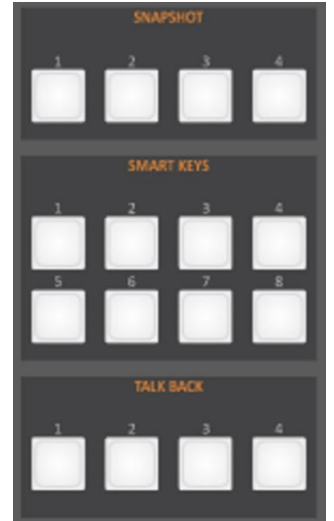
- **K1-8**

These buttons allow the console to send commands to compatible external software or equipment through the network. The supported communication methods include:

- **TCP**
- **UDP**
- **REST API**

A Smart Key button may therefore be used to control:

- Radio-automation software
- Audio players
- Recording systems
- External codecs
- Studio-control software
- Lighting systems
- Streaming equipment
- Remote audio devices
- Broadcast processors
- Other network-compatible systems



Each Smart Key must be configured with the required network command and destination. Pressing the button then transmits the assigned command without requiring the operator to open a separate control application. The Smart Keys can operate in two principal modes:

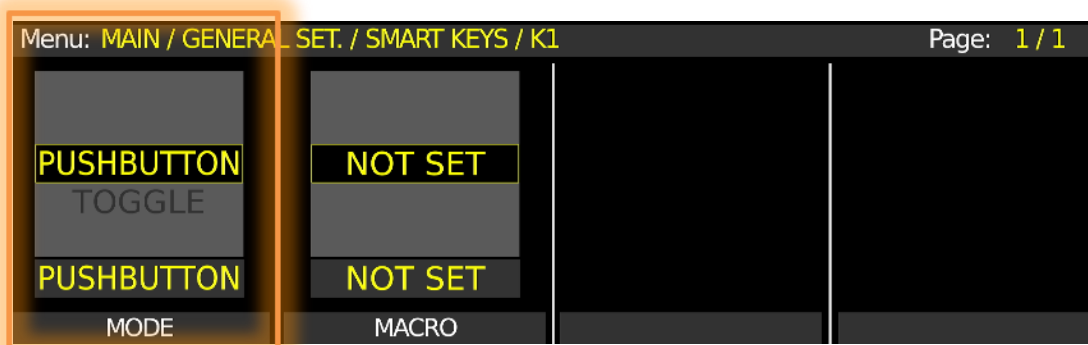
- **PUSHBUTTON**
- **TOGGLE**

PUSHBUTTON:

In **IMPULSIVE** mode, the assigned command is generated by a single button press. The button behaves like a momentary command. It performs the assigned operation when pressed without maintaining a permanent ON/OFF state. This mode may be used for commands such as:

- Start an audio player
- Stop a recording
- Recall a remote function
- Send a cue
- Activate a temporary external command
- Execute a software macro

The exact duration and behaviour of the command depend on the Smart Key configuration and the controlled system.



TOGGLE:

In **TOGGLE** mode, the Smart Key alternates between two operating states. The normal sequence is:

1. The first button press activates the assigned ON state or command.
2. The button remains logically active.

3. The second button press activates the OFF state or corresponding command.
4. The button returns to its inactive condition.

TOGGLE may be used for functions such as:

- Enabling or disabling an external process
- Starting and stopping a continuous function
- Activating and deactivating a Studio light
- Switching an external device between two states
- Enabling or disabling a remote software function

The button illumination may provide a visual indication of its current state, depending on the console and Smart Key configuration. **Important:** FUNCTIONS buttons transmit commands but do not confirm that the external device has completed the requested operation unless a corresponding feedback system has been configured.

SMART KEYS must be distinguished from the channel F1 and F2 buttons:

- **FUNCTIONS K1–K8** are global Smart Keys principally used to control external applications or devices.
- **F1 and F2** are configurable controls associated with an individual channel and its selected source.

SMART KEYS must also be distinguished from **ONAIR TRIGGERS**. Smart Keys send commands when the operator presses K1–K8, while ONAIR TRIGGERS can send commands automatically when a channel fader or ON/START control changes state.

Warning: verify every Smart Key command before live operation. An incorrect IP address, port, protocol or command string can operate the wrong device or fail to produce the expected result.



TALKBACK:

The **TALKBACK** section provides four direct buttons used to activate the console's Talkback communication circuits. Talkback allows the Control Room operator, producer or technician to communicate with:

- Presenters
- Guests
- Studio technicians
- Producers
- Assistants
- Personnel in another Studio
- Other configured communication destinations

When a TALKBACK button is pressed, the configured Talkback microphone is routed to the destination associated with that button. Depending on the installed firmware and system configuration, the buttons may correspond to different Talkback buses or destinations, such as:

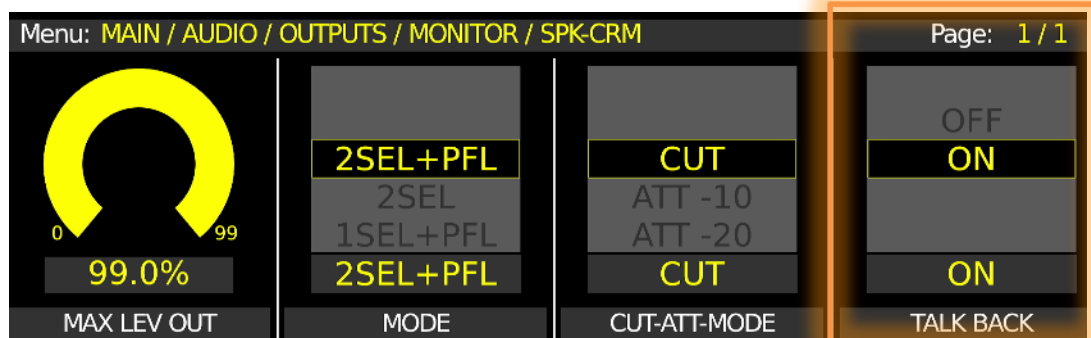
- **TB1**
- **TB2**
- **TB3**
- **TB4**

The exact destination of each physical TALKBACK button depends on:

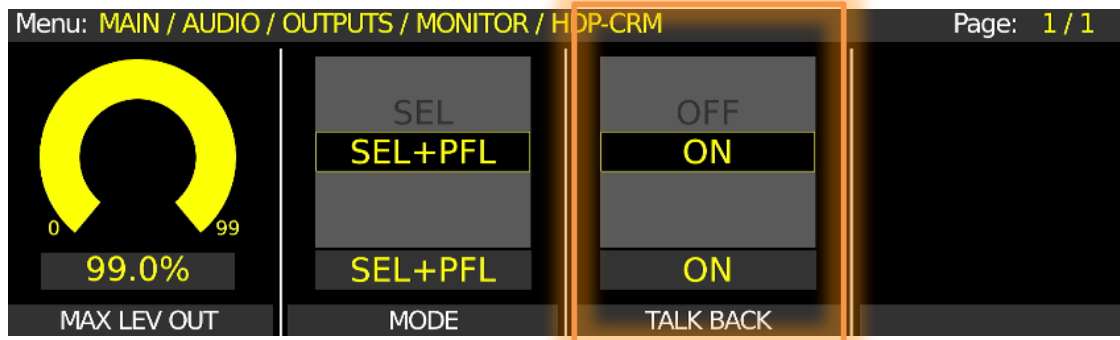
- Talkback microphone configuration
- Talkback-bus assignments
- Studio monitor settings
- Headphone monitor settings
- Installed firmware
- External Talk Box configuration
- System routing



The Talkback communication received through the Studio monitor paths can normally be configured through menus such as: **MAIN → AUDIO → OUTPUTS → MONITOR → SPK-CRM → TALKBACK**



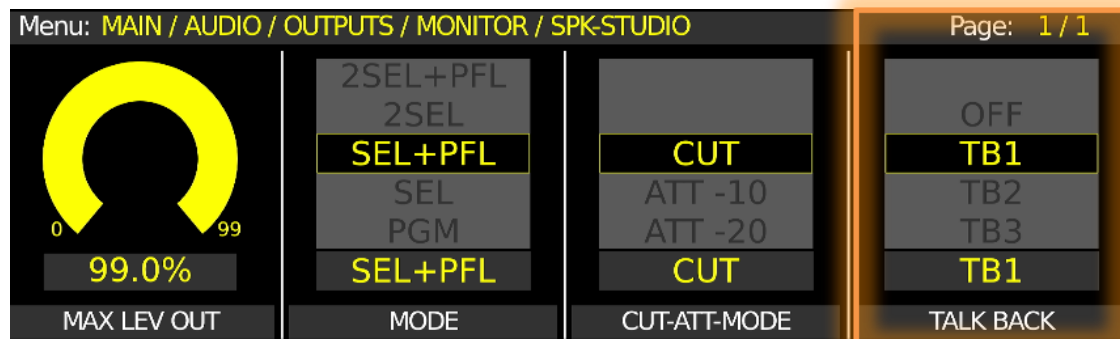
or: **MAIN → AUDIO → OUTPUTS → MONITOR → HDP-CRM → TALKBACK**



or: **MAIN → AUDIO → OUTPUTS → MONITOR → HDP-STUDIO → TALKBACK**



or: **MAIN → AUDIO → OUTPUTS → MONITOR → SPK-STUDIO → TALKBACK**



The physical TALKBACK button activates the communication command. It does not independently configure:

- Which microphone is used for Talkback
- The Talkback microphone gain
- The receiving monitor output
- The Talkback destination
- The Talkback level
- Whether programme audio is mixed, attenuated or interrupted
- The Studio loudspeaker CUT or attenuation behaviour

These functions must be configured through the corresponding input, monitoring and Talkback menus.

Talkback can remain available through Studio headphones even when the Studio loudspeakers are muted because a microphone is active. This allows the Control Room operator to communicate with the presenter

without creating acoustic feedback. **Important:** verify the destination associated with each TALKBACK button before using it. A command sent to the wrong Talkback bus may be heard by unintended personnel or may fail to reach the required Studio.

Warning: Talkback is normally intended for internal communication and may contain private or technical information. Confirm that the Talkback signal is not unintentionally routed to PGM, SUB, AUX or another On Air output.

MONITOR SECTION

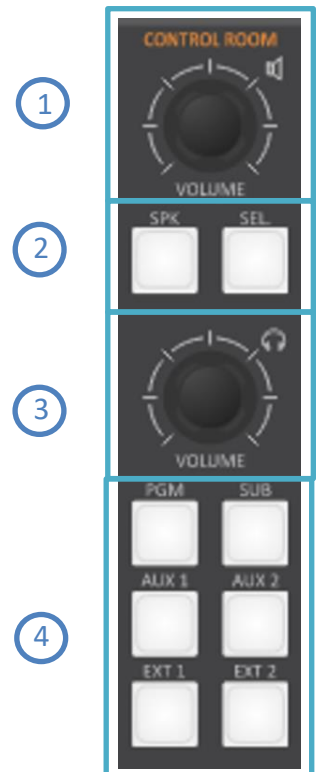
STUDIO & CONTROL ROOM:

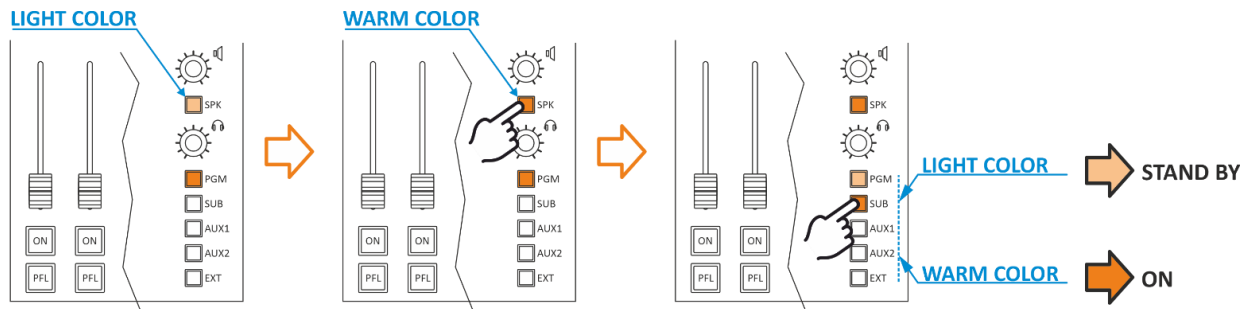
- This section is used for the management of the **Control Room** and **Studio** loudspeakers and Headphones.
- The rotary control knobs allow you to **amplify/attenuate** the audio level.
- The audio level goes from **0** to **99** is the maximum allowed level. Can set the maximum level by going to this page:
MAIN → AUDIO → OUTPUTS → MONITOR → SPK-STUDIO
- The step of the loudspeaker adjustment is **1 dB** and the level goes from **-80 dB** to the maximum of **+19 dB**.
(lev.0=-80dBu, lev. 80=0dBu, lev.99=+19dBu)
- By **pressing the knob**, you can mute  or play the control room monitors if it is already muted.
- To unmute the speaker  just press the knob or **increase/decrease** the audio level by rotating the knob and confirm that by clicking on the same knob.
- The loudspeaker's level and status are displayed in the bottom-right section of the LCD.



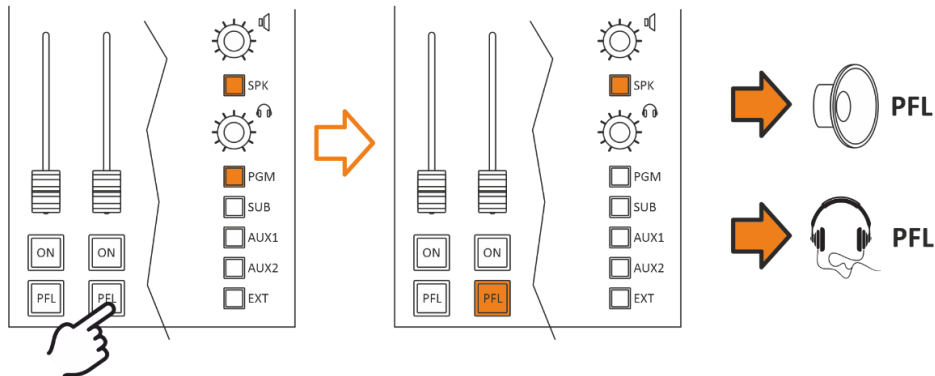
MODE: 1SEL, 1SEL+PFL, 2SEL or 2SEL+PFL

- **PFL (pre-fader listen):** This mode allows you to listen in speakers to the audio of the single channel **before** the intervention of the fader.
- **1SEL (one selection):** This option allows you to listen in speakers to only ONE selected output (the same of the headphones) from the output section  (PGM, SUB, AUX1, AUX2, EXT).
- **1SEL+PFL:** This mode allows you to listen to **ONE selected output** or **PFL** if press the button  PFL of one channel.
- **2SEL (two selections):** This option allows you to listen to ONE selected output  (PGM, SUB, AUX1, AUX2, EXT), And by pressing the **SPK** button  you will be able to listen to different outputs in speakers. (Headphones and Speaker can listen different sources)
- **EX.:** if you need to listen to different output in **SPK-CRM** Control Room Speakers, select the **2SEL** mode from the setting and press the **SPK** button in section  (it will show up in warm color), then press any other output button from section  (it will show up in warm color) to hear that output in the speakers only. (see the next figure).

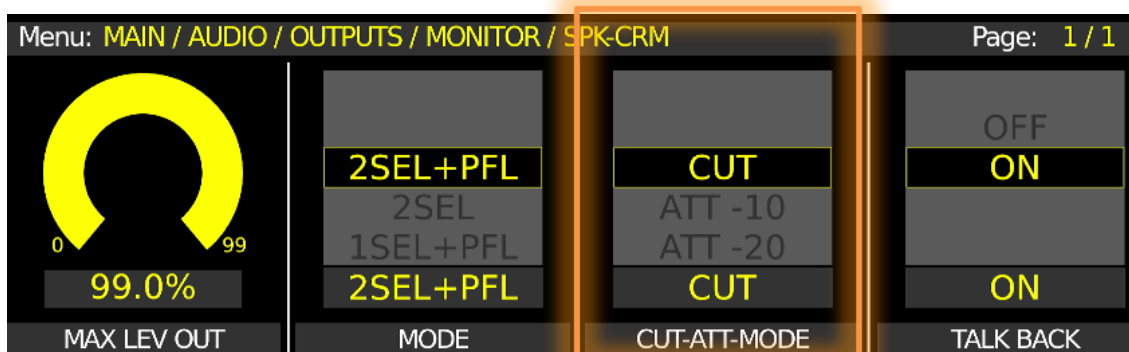




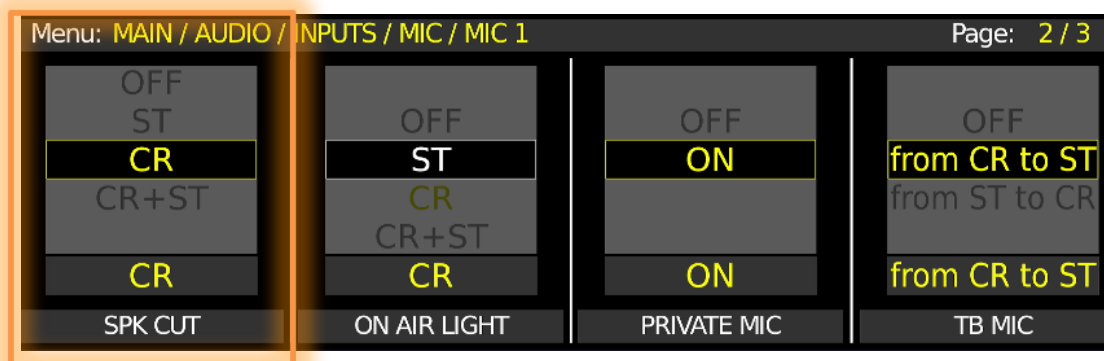
- **2SEL+PFL:** It is a similar mode to the above-mentioned case, in addition to that you can hear the PFL in a bout of headphones and speakers simultaneously. (see the next figure).



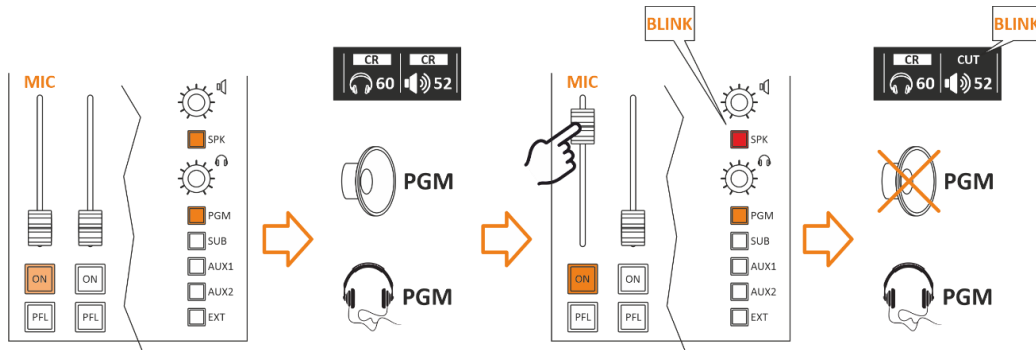
CUT-ATT-MODE:



CUT MODE: it's allowed the microphone to **CUT OFF** the audio of control room loudspeakers once the microphone goes to ON. The opening of a microphone channel (if configured appropriately) can generate the closing command of the loudspeakers. That is possible to choose one or more microphones to cut the loudspeaker output of the **CR** "Control Room" by following this path: **MAIN → AUDIO → INPUTS → MIC → MIC 1.**



When you select **CR**, press down the knob to confirm the selection and the **CR** will show up in yellow color.



When you open the associated source with a **CUT** function you will see it will **MUTE** CR Loudspeakers output to prevent LOOP audio and the **SPEAKER** icon at LCD will start plinking.

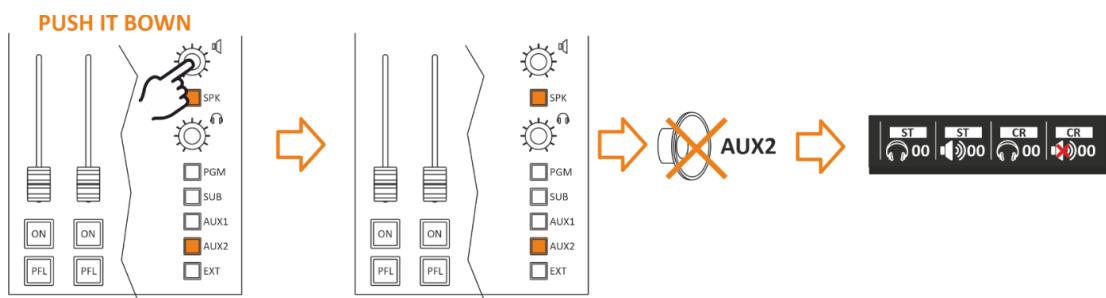
ATT. (attenuation) MODE: It helps reduce acoustic flux from flowing into the speakers. Is it possible to decrease the output of the loudspeaker to **-40, -30, -20, or -10 dB**.

EX.: If we want to reduce the sound coming out of the speakers by **40 dB** less than the current value, then we choose **-40**. The same thing if we want to reduce the current volume when opening any of the microphones with a value of **10** decibels, then we choose **-10** and confirm the selection by pressing the button around until the selection color changes to yellow.

The **CUT** mode is triggered by the change from **OFF** to **ON** of a microphone source to which it has been set closing of the loudspeaker.

As shown in the MENU this function (**CUT**) is associated only with the loudspeakers, to avoid LARSEN effects “feedback loop” from occurring between the nearby loudspeakers and On-Air microphones.

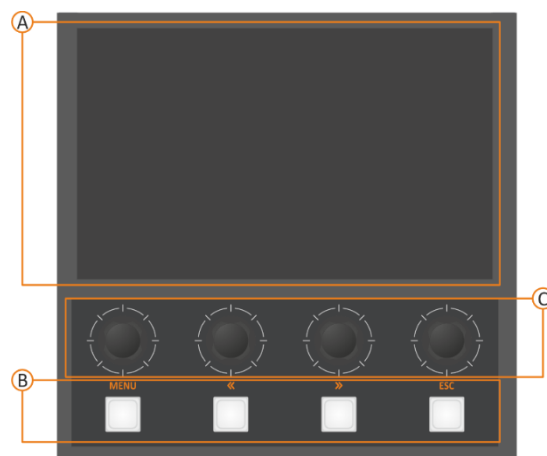
On the other hand, if you need to **MUTE** the loudspeakers manually just **PRESS** the volume knob down. **PRESS** the volume knob a second time or rotate it to activate the loudspeaker output and amplification or attenuation. The status of **MUTE-SPK** is indicated by a red cross on the **SPEAKER** icon.



TALKBACK:

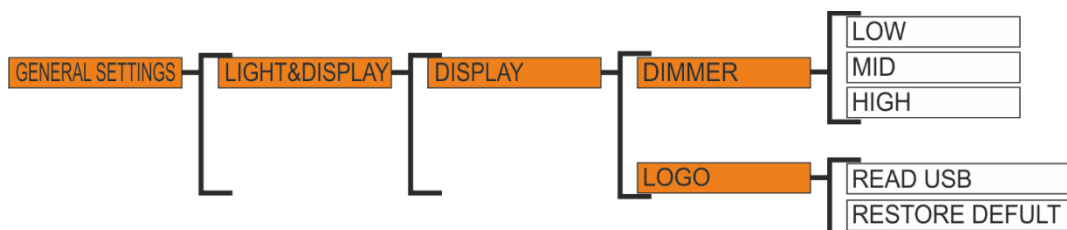
Here is where can **disable/enable** the **TALKBACK** communications in the loudspeaker’s output. Normally the talkback is used to communicate between the mixer man and the people in the studio via headphones. This option gives you the possibility to hear the people talking in **TALKBACK** the studio via the **CR** (Control Room loudspeakers).

MENU CONTROL (BUTTONS & DISPLAY 7" TFT)



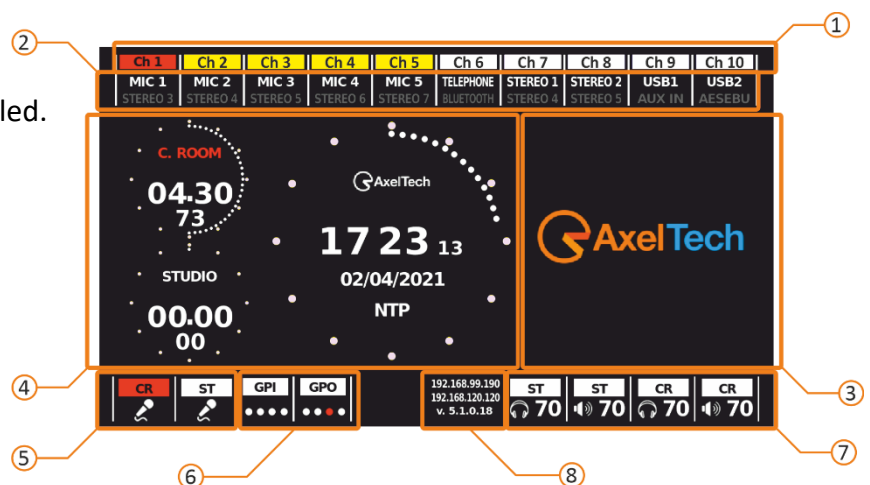
COLOR DISPLAY 7" - 800X480 RESOLUTION:

The display is used to control the mixer setting and monitor all the parameters and it has its own configuration menu. The display has some parameters that can be changed like **DIMMER** and **LOGO**. It is possible to adjust the display brightness, allowing you to deal with any environmental situation. You can find the parameter in the following menu: **GENERAL SETTINGS/LIGHT & DISPLAY/DISPLAY/DIMMER** (low, mid, high). It is possible to have the logo of your own station on the Mixer display. The image must be 330x280px in **PNG** format.

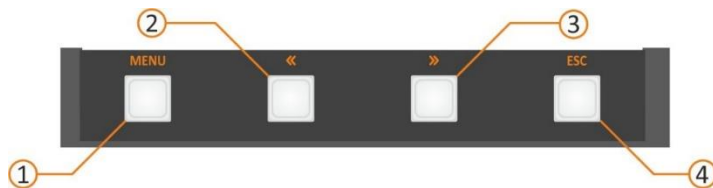


The **HOME THEME** parameter defines what to see on the display in standard conditions when no function has been recalled. The following picture is the **HOME PAGE**:

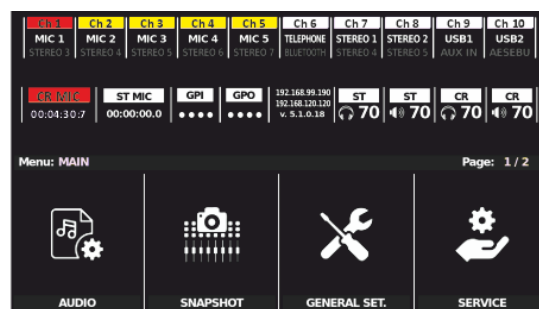
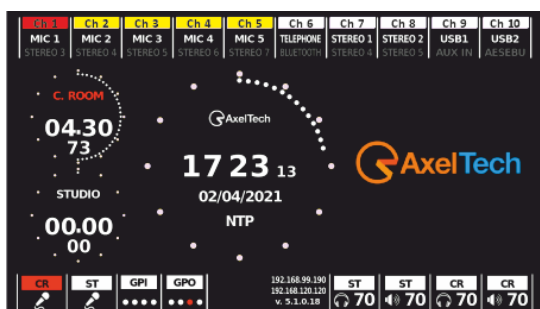
1. **OFF / STAND-BY / ON-AIR** channel STATUS.
2. Source names associated with channels.
3. Logo display.
4. NTP synchronized clock & Mic timer.
5. ONAIR mic status.
6. GPIO 1&2 status.
7. Speakers and Headphones status.
8. Firmware release and IP address.



BUTTONS FOR THE MENU NAVIGATION:

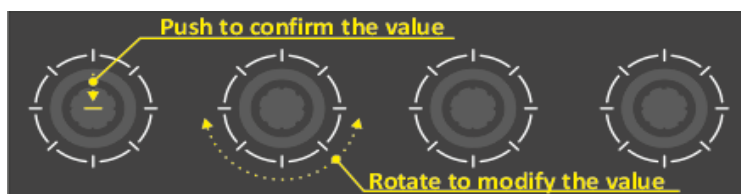


- 1) The **MENU** button switches from HOME to MENU. With this button, you can recall the mixer configuration menu.



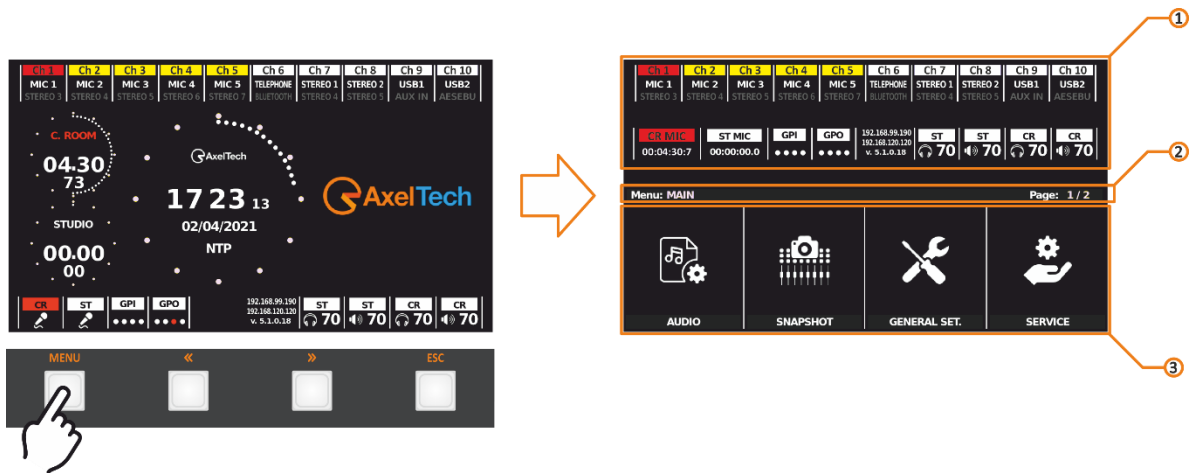
Buttons 2 and 3 (>> and <<) allow you to move between pages of **parameters/functions** on the same menu level. **ESC (button 4)** allows you to exit from the menu level going up in the menu.

ENCODER FOR THE MENU NAVIGATION:



- The 4 knobs allow you to adjust and confirm the menu parameters.
- It is also possible to **PUSH** the knob to confirm the parameter setting.
- Menus are designed to have a perfect match between graphics and knobs.
- Parameter changes will be directly applied to the audio of the aired channel.
- The yellow value represents the stored value.
- While a knob is being moved the circle becomes white. A white value appears in the middle of the circle. The new value represents an instantaneous value that is different from the stored one.
- Push the related knob to confirm the parameter changes. Once you have confirmed the new value, the graphic will become yellow again.
- At the **ESC** press or at the next screen change if the changes will not be confirmed, the value will be restored to the one previously saved.

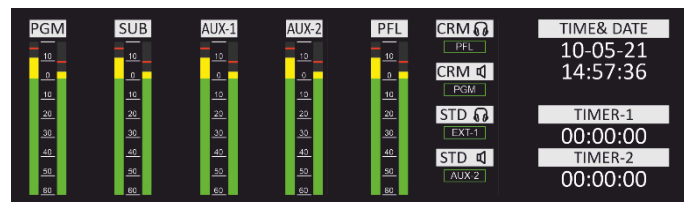
MENU display parts



1. In this part of the display can see some useful elements and information depending on the location of the MENU.
2. From here you can browse all lists available in the menu.
3. To navigate the root of your page. The page number indicates the pages available and the current page.

OUTPUTS DISPLAY

A built-in LCD audio level bar graph meter.



The LCD audio level bar graph meter would display the audio level for each of the mixer outputs, including PGM, SUB, AUX1, AUX2, and PFL. This would allow the operator to easily monitor the level of each output and make adjustments as needed to ensure that the audio is properly balanced and not distorted.

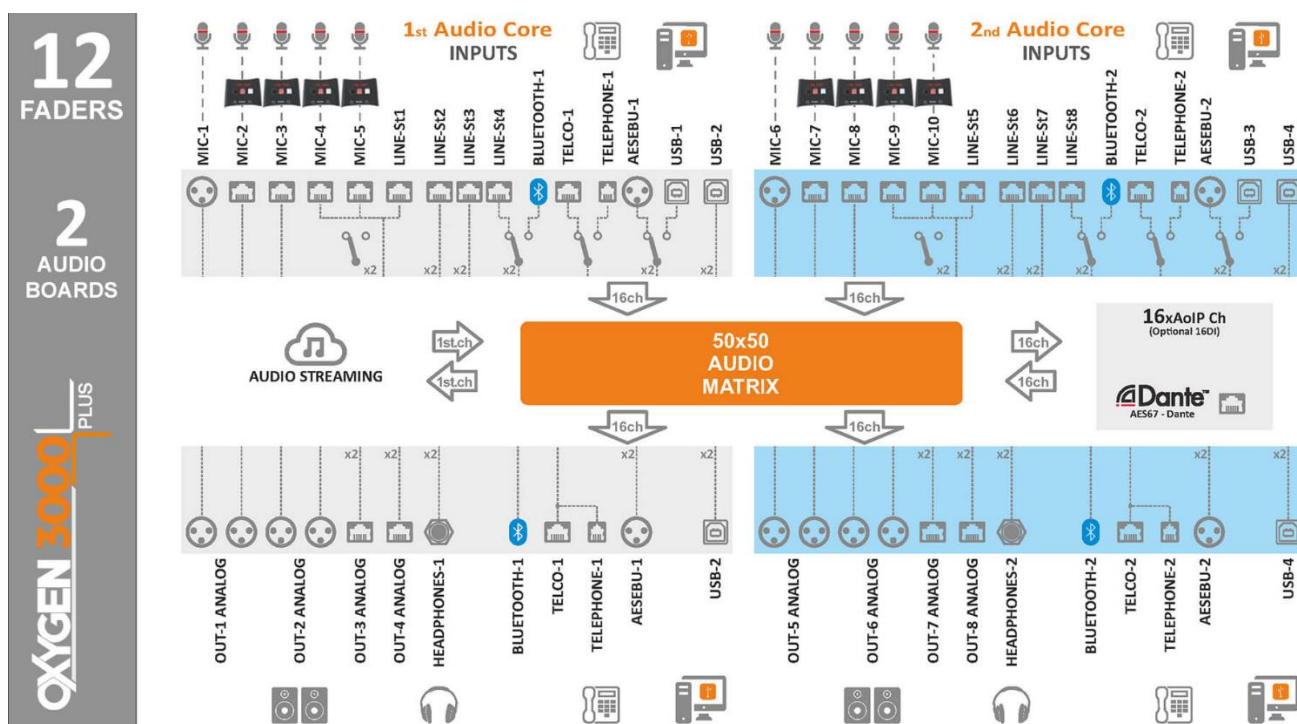
It also contains the status of **audio monitoring** devices, such as loudspeakers and headphones. This would allow the operator to control the volume of the monitors and audio routing.

In addition, on the far right side are displayed some details about the date, time, and **2 TIMERS**. This information could be useful for keeping track of recording sessions or live events and ensuring that everything is running on schedule.

MENU

INPUT MODE CONFIGURATION

Before selecting any of the operating modes described below, the corresponding input resources of the Oxygen 3000 Plus must first be configured in the SETTINGS section. Each selectable mode depends on the hardware input assignment previously established for the relevant Audio Core. Therefore, the required input configuration must be selected before the associated source or operating mode can be made available on the mixer.



MIC / STEREO INPUT SWITCH:

1st Audio Board: MIC-4 + MIC-5 / LINE-ST1

2nd Audio Board: MIC-9 + MIC-10 / LINE-ST5

Selects whether the corresponding pair of analogue input channels operates as **two independent mono microphone inputs** or as **one balanced stereo line input**.

- **2 MIC MODE:** enables MIC-4 and MIC-5 on the first Audio Core, or MIC-9 and MIC-10 on the second Audio Core. The corresponding stereo line input is unavailable.
- **STEREO MODE:** combines the two input channels into LINE-ST1 or LINE-ST5, providing a Left/Right stereo input. The corresponding microphone inputs are unavailable.

This switch allows each Audio Core to provide either **five microphone inputs and fewer stereo inputs** or **three microphone inputs and one additional stereo input**.

ATTENTION: To configure the **input assignment**, open the **INPUT/OUTPUT MODE** interface under **SETTINGS**. For further information, refer to the **INPUT/OUTPUT MODE** section on the following pages.

LINE / BLUETOOTH INPUT SWITCH:

1st Audio Board: LINE-ST4 / BLUETOOTH-1

2nd Audio Board: LINE-ST8 / BLUETOOTH-2

Selects whether the corresponding stereo resource is assigned to an **analogue line input** or to the built-in Bluetooth module.

- **LINE STEREO MODE:** enables LINE-ST4 or LINE-ST8 as a balanced stereo analogue input. Bluetooth audio is unavailable on that Audio Core.
- **BLUETOOTH MODE:** enables BLUETOOTH-1 or BLUETOOTH-2. The corresponding LINE-ST input becomes unavailable.

Bluetooth can operate bidirectionally for telephone calls or as a high-quality stereo source for audio received from a smartphone, tablet or computer.

ATTENTION: To configure the **input assignment**, open the **INPUT/OUTPUT MODE** interface under **SETTINGS**. For further information, refer to the **INPUT/OUTPUT MODE** section on the following pages.

TELCO / TELEPHONE INPUT SWITCH:

1st Audio Board: TELCO-1 / TELEPHONE-1

2nd Audio Board: TELCO-2 / TELEPHONE-2

Selects the type of telephone interface assigned to the Audio Core.

- **TELCO MODE:** enables the connection to an **external telephone hybrid**. The console manages the receive audio, N-1 return signal and the associated Ring/Hook commands through GPIO or network control.
- **TELEPHONE MODE:** enables the Oxygen 3000 Plus **internal POTS/PSTN telephone hybrid**, connected through the corresponding LINE and SET ports.

Only one of the two telephone interfaces can use the assigned resource at a time.

ATTENTION: To configure the **input assignment**, open the **INPUT/OUTPUT MODE** interface under **SETTINGS**. For further information, refer to the **INPUT/OUTPUT MODE** section on the following pages.

AES/EBU / USB INPUT SWITCH:

1st Audio Board: AES/EBU-1 / USB-1

2nd Audio Board: AES/EBU-2 / USB-3

Selects whether the corresponding digital input resource receives audio from the professional AES/EBU interface or from a USB playback channel.

- **AES/EBU MODE:** enables the stereo **AES3 digital input** through the XLR connector. The input supports external digital equipment such as audio processors, codecs and professional players.
- **USB MODE:** enables the corresponding stereo USB playback input, allowing audio to be received directly from a connected computer.

When AES/EBU is selected, the USB input located on the opposite side of the switch is unavailable as a playback source, and vice versa.

ATTENTION: To configure the **input assignment**, open the **INPUT/OUTPUT MODE** interface under **SETTINGS**. For further information, refer to the **INPUT/OUTPUT MODE** section on the following pages.

SETTINGS configures the general operating behaviour of the console's audio system and physical control surface. This section includes:

- INPUT MODE
- SURFACE MODE
- FADER SETTINGS
- VJ PRO MODE

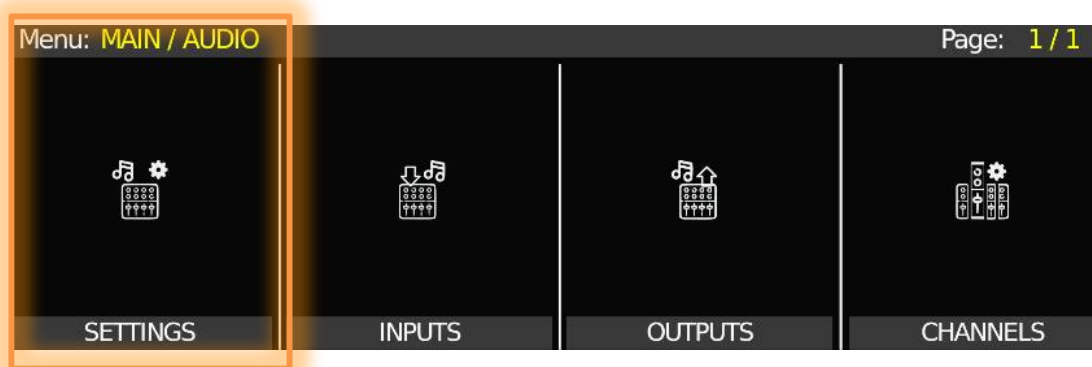
AUDIO

The AUDIO menu, located under MAIN, is the principal section used to configure the audio operation of the OXYGEN 3000 Plus. From this menu, the operator can:

- Configure the general behaviour of the audio system
- Select the operating mode of the physical inputs
- Configure every available input source
- Assign and adjust the console outputs
- Define monitoring and Talkback behaviour
- Assign sources to physical channels
- Configure the primary and secondary source of each channel
- Enable additional virtual channels
- Adjust audio processing such as EQ, compression and ducking
- Configure signal routing to PGM, SUB, AUX1 and AUX2

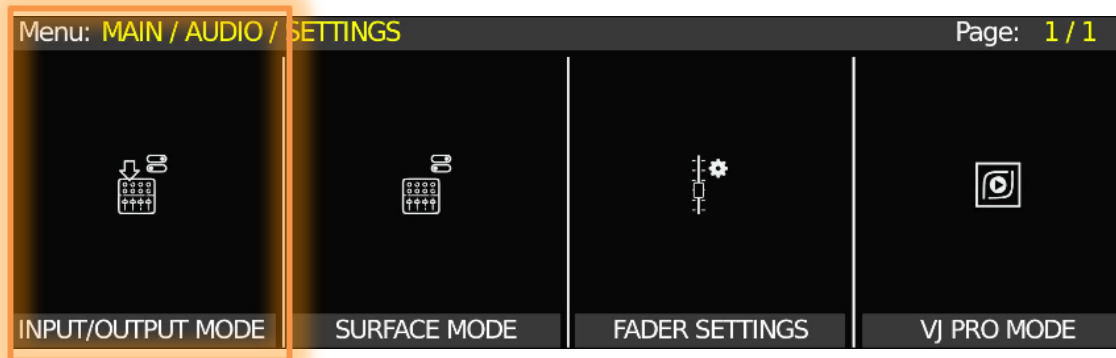
Important: parameters configured inside AUDIO can directly affect the signals currently sent On Air. Changes should therefore be performed carefully, especially when modifying input gain, routing, processing or output assignments.

1. SETTINGS



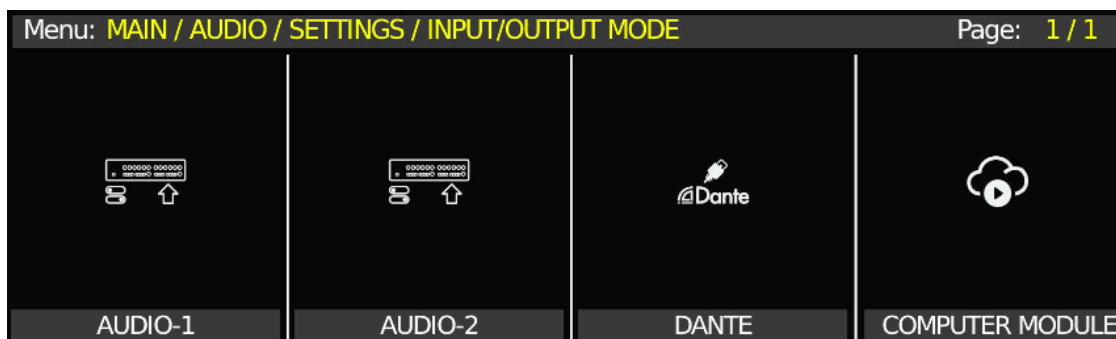
1.1 INPUT/OUTPUT MODE

The **INPUT/OUTPUT MODE** menu, located under **SETTINGS**, defines how the physical and logical audio connections of the OXYGEN 3000 Plus are presented to the internal routing system. It can be accessed through: **MAIN → AUDIO → SETTINGS → INPUT/OUTPUT MODE**



The section includes:

- **AUDIO-1**
- **AUDIO-2**
- **DANTE**
- **COMPUTER MODULE**



AUDIO 1-2 INPUT



The **AUDIO-1 INPUT** and **AUDIO-2 INPUT** pages, located under **INPUT/OUTPUT MODE**, define how the input resources of the first and second Audio I/O Boards are presented to the OXYGEN 3000 Plus routing system. They can be accessed through:

MAIN → AUDIO → SETTINGS → INPUT/OUTPUT MODE → AUDIO-1 → INPUT

or, when a second Audio I/O Board is installed:

MAIN → AUDIO → SETTINGS → INPUT/OUTPUT MODE → AUDIO-2 → INPUT

The OXYGEN 3000 Plus supports one or two Audio I/O Boards. AUDIO-1 represents the first board, while AUDIO-2 represents the optional second board and uses the same configuration logic with the corresponding input numbers.

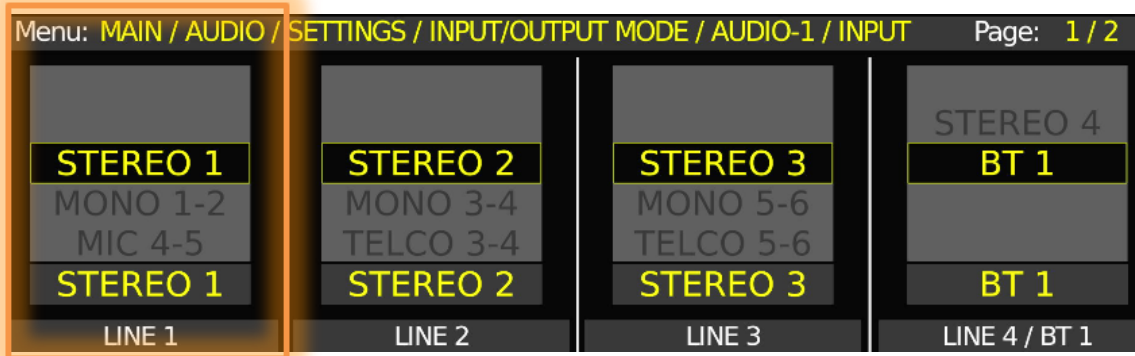
ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

-AUDIO-1 INPUT:

AUDIO-1 INPUT configures the inputs belonging to the first Audio I/O Board. The page includes:

- **LINE 1**
- **LINE 2**
- **LINE 3**
- **LINE 4 / BT 1**
- **USB 1 / DIGITAL 1**
- **TEL 1 / TELCO 1**

LINE 1:



LINE 1 is a combined analogue connection that can operate as a stereo line input, two independent mono line inputs or two microphone inputs. The selectable options are:

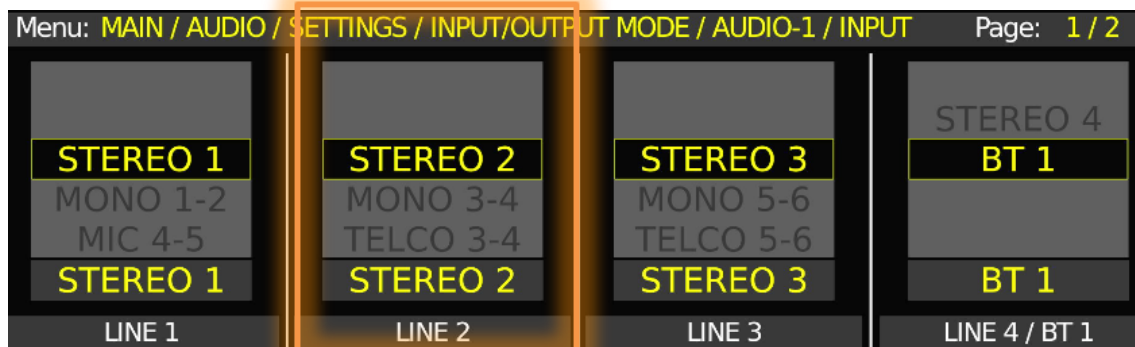
- **STEREO**
- **2 MONO**
- **2 MIC**

STEREO: LINE 1 is presented as one Left/Right line-level source.

2 MONO: LINE 1 is divided into two independent mono line-level sources.

2 MIC: The connection is assigned to two additional microphone sources and enables the microphone-specific parameters supported by the hardware.

Important: changing LINE 1 to 2 MIC removes the corresponding LINE 1 source. Existing channel assignments using LINE 1 must therefore be checked.

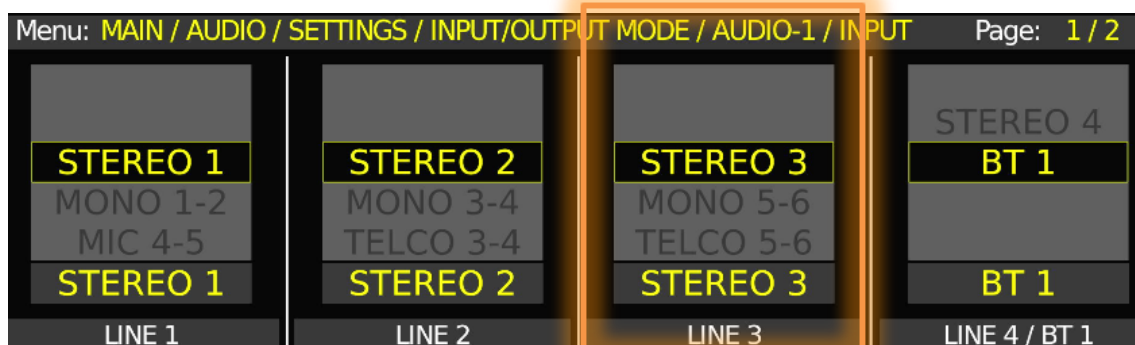
LINE 2:

LINE 2 defines the structure of the second analogue line-input pair. The selectable options are:

- **STEREO**
- **2 MONO**
- **2 TELCO**

LINE 2 uses the same operating logic as LINE 1:

- **STEREO:** one Left/Right source
- **2 MONO:** two independent mono sources
- **2 TELCO:** two independent telephone-interface sources.

LINE 3:

LINE 3 defines the structure of the third analogue line-input pair. The selectable options are:

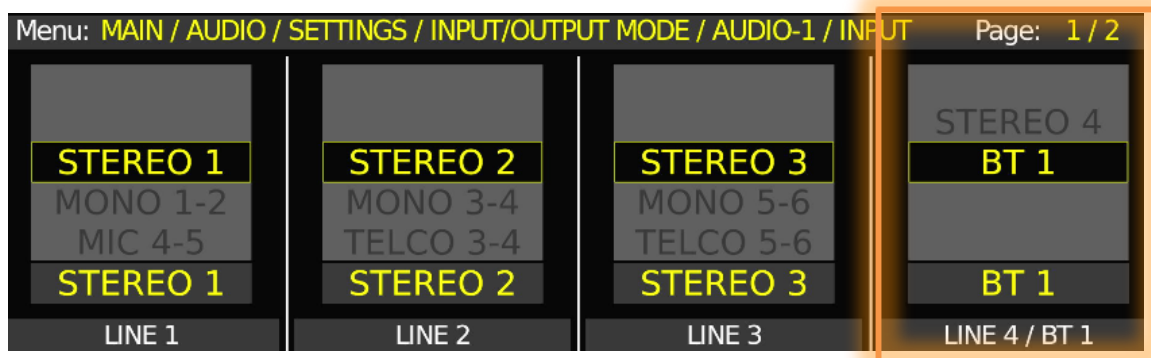
- **STEREO**
- **2 MONO**
- **2 TELCO**

LINE 3 uses the same operating logic as LINE 2:

- **STEREO:** one Left/Right source
- **2 MONO:** two independent mono sources
- **2 TELCO:** two independent telephone-interface sources

LINE 2 and LINE 3 can be configured differently. For example, LINE 2 may remain stereo while LINE 3 is divided into two mono or TELCO sources.

LINE 4 / BT 1:



LINE 4 / BT 1 determines whether the shared input resource of the first board is used as an analogue line input or as the first Bluetooth input. The selectable options are:

- **STEREO**
- **2 MONO**
- **2 TELCO**
- **BT 1**

LINE 4 uses the same operating logic as LINE 3:

- **STEREO:** one Left/Right source
- **2 MONO:** two independent mono sources
- **2 TELCO:** two independent telephone-interface sources
- **BT 1:** The analogue LINE 4 resource is replaced by the first Bluetooth interface.

The actual Bluetooth behaviour depends on the connected device and on the Bluetooth configuration.

USB 1 / DIGITAL 1:



USB 1 / DIGITAL 1 selects which digital input resource is used by the first Audio I/O Board.

The selectable options are:

- **USB 1**
- **DIGITAL 1**

USB 1: The console receives audio from a connected computer through the first USB audio interface.

The OXYGEN 3000 Plus is detected by the computer as an audio device, allowing compatible software to send programme audio to the console. USB 1 is normally presented as a stereo input source.

DIGITAL 1: The console receives audio from the first physical digital input instead of the USB playback input. The physical digital input uses the board's balanced digital connection and is normally presented as a stereo digital source.

Important: USB 1 and DIGITAL 1 are alternative uses of the same input resource. Only the selected source is made available to the internal routing system.

TEL 1 / TELCO 1:



TEL 1 / TELCO 1 determines whether the first board uses its internal telephone hybrid or an external TELCO interface. The selectable options are:

- **TEL 1**
- **TELCO 1**

TEL 1: TEL 1 enables the internal telephone hybrid associated with the first Audio I/O Board. The related F1 or F2 button can be configured for telephone operation.

TELCO 1: TELCO 1 selects the interface intended for an external telephone hybrid, codec or communication device. TELCO 1 is used when the telephone-line interface is handled by external equipment rather than by the console's internal hybrid.

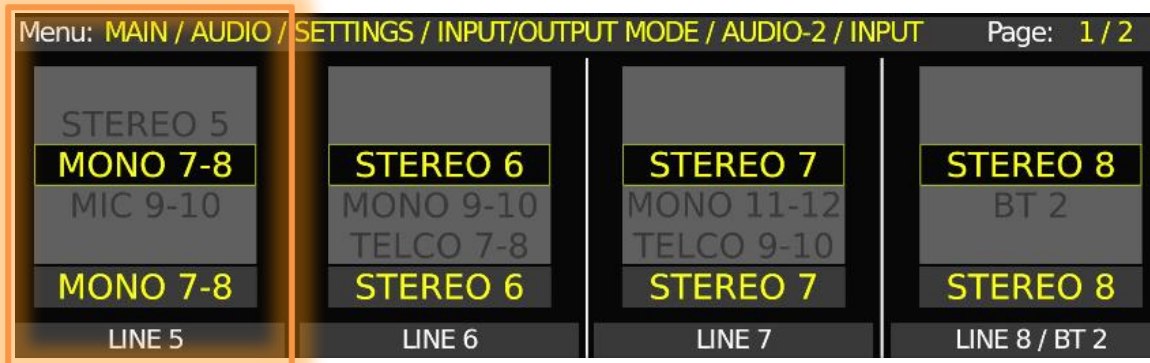
Important: TEL 1 and TELCO 1 are mutually exclusive. Selecting TELCO 1 does not automatically configure the external hybrid, N-1 signal or GPI/GPO assignments.

-AUDIO-2 INPUT

AUDIO-2 INPUT configures the inputs belonging to the first Audio I/O Board. The page includes:

- **LINE 5**
- **LINE 6**
- **LINE 7**
- **LINE 8 / BT 2**
- **USB 3 / DIGITAL 2**
- **TEL 2 / TELCO 2**

LINE 5:



LINE 5 is a combined analogue connection that can operate as a stereo line input, two independent mono line inputs or two microphone inputs. The selectable options are:

- **STEREO**
- **2 MONO**
- **2 MIC**

STEREO: LINE 5 is presented as one Left/Right line-level source.

2 MONO: LINE 5 is divided into two independent mono line-level sources.

2 MIC: The connection is assigned to two additional microphone sources and enables the microphone-specific parameters supported by the hardware.

Important: changing LINE 5 to 2 MIC removes the corresponding LINE 5 source. Existing channel assignments using LINE 5 must therefore be checked.

LINE 6:



LINE 6 defines the structure of the second analogue line-input pair. The selectable options are:

- **STEREO**
- **2 MONO**
- **2 TELCO**

LINE 6 uses the same operating logic as LINE 5:

- **STEREO:** one Left/Right source
- **2 MONO:** two independent mono sources
- **2 TELCO:** two independent telephone-interface sources

LINE 7:

LINE 7 defines the structure of the third analogue line-input pair. The selectable options are:

- **STEREO**
- **2 MONO**
- **2 TELCO**

LINE 7 uses the same operating logic as LINE 6:

- **STEREO:** one Left/Right source
- **2 MONO:** two independent mono sources
- **2 TELCO:** two independent telephone-interface sources

LINE 6 and LINE 7 can be configured differently. For example, LINE 6 may remain stereo while LINE 7 is divided into two mono or TELCO sources.

LINE 8 / BT 2:

LINE 8 / BT 2 determines whether the shared input resource of the first board is used as an analogue line input or as the first Bluetooth input. The selectable options are:

- **STEREO**
- **2 MONO**
- **2 TELCO**
- **BT 2**

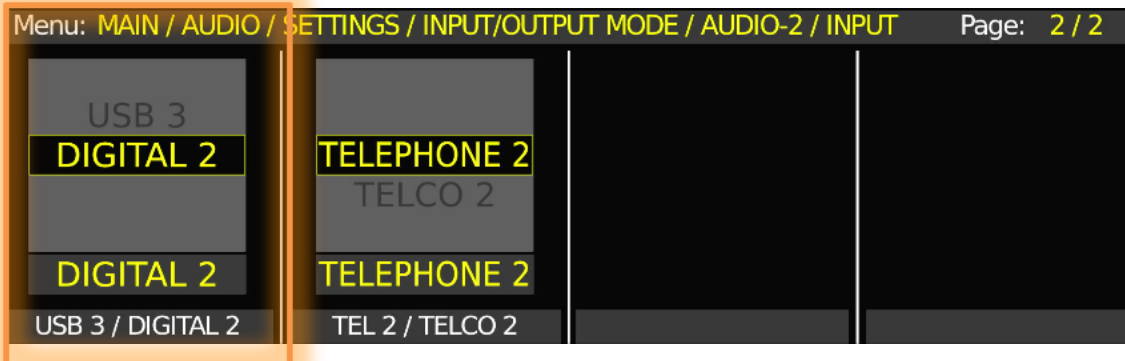
LINE 4 uses the same operating logic as LINE 7:

- **STEREO:** one Left/Right source
- **2 MONO:** two independent mono sources
- **2 TELCO:** two independent telephone-interface sources

- **BT 2:** The analogue LINE 4 resource is replaced by the first Bluetooth interface.

The actual Bluetooth behaviour depends on the connected device and on the Bluetooth configuration.

USB 3 / DIGITAL 2:



USB 3 / DIGITAL 2 selects which digital input resource is used by the first Audio I/O Board. The selectable options are:

- **USB 3**
- **DIGITAL 2**

USB 3: The console receives audio from a connected computer through the first USB audio interface. The OXYGEN 3000 Plus is detected by the computer as an audio device, allowing compatible software to send programme audio to the console. USB 1 is normally presented as a stereo input source.

DIGITAL 2: The console receives audio from the first physical digital input instead of the USB playback input. The physical digital input uses the board's balanced digital connection and is normally presented as a stereo digital source.

Important: USB 1 and DIGITAL 1 are alternative uses of the same input resource. Only the selected source is made available to the internal routing system.

TEL 2 / TELCO 2:



TEL 2 / TELCO 2 determines whether the first board uses its internal telephone hybrid or an external TELCO interface. The selectable options are:

- **TEL 2**
- **TELCO 2**

TEL 2: TEL 1 enables the internal telephone hybrid associated with the first Audio I/O Board. The related F1 or F2 button can be configured for telephone operation.

TELCO 2: TELCO 2 selects the interface intended for an external telephone hybrid, codec or communication device. TELCO 1 is used when the telephone-line interface is handled by external equipment rather than by the console's internal hybrid.

Important: TEL 2 and TELCO 2 are mutually exclusive. Selecting TELCO 2 does not automatically configure the external hybrid, N-1 signal or GPI/GPO assignments.

AUDIO 1-2 OUTPUT

The **AUDIO-1 OUTPUT** and **AUDIO-2 OUTPUT** pages, located under **INPUT/OUTPUT MODE**, define how the analogue outputs of each Audio I/O Board are presented to the OXYGEN 3000 Plus routing system.

They can be accessed through:

MAIN → AUDIO → SETTINGS → INPUT/OUTPUT MODE → AUDIO-1 → OUTPUT

or, when a second Audio I/O Board is installed:

MAIN → AUDIO → SETTINGS → INPUT/OUTPUT MODE → AUDIO-2 → OUTPUT

Each Audio I/O Board provides four balanced analogue output positions:

- **ANALOG 1/5**
- **ANALOG 2/6**
- **ANALOG 3/7**
- **ANALOG 4/8**

The first two outputs use balanced XLR connections, while ANALOG 3/7 and ANALOG 4/8 use balanced RJ45 connections. The OXYGEN 3000 Plus therefore provides four assignable analogue outputs for each installed Audio I/O Board.

Important: AUDIO-1/2 OUTPUT determines the fundamental structure of the physical output. It does not select which audio signal is sent to that output. PGM, SUB, AUX, monitor and direct signals are assigned separately under **MAIN → AUDIO → OUTPUTS**.

ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

-AUDIO-1 OUTPUT

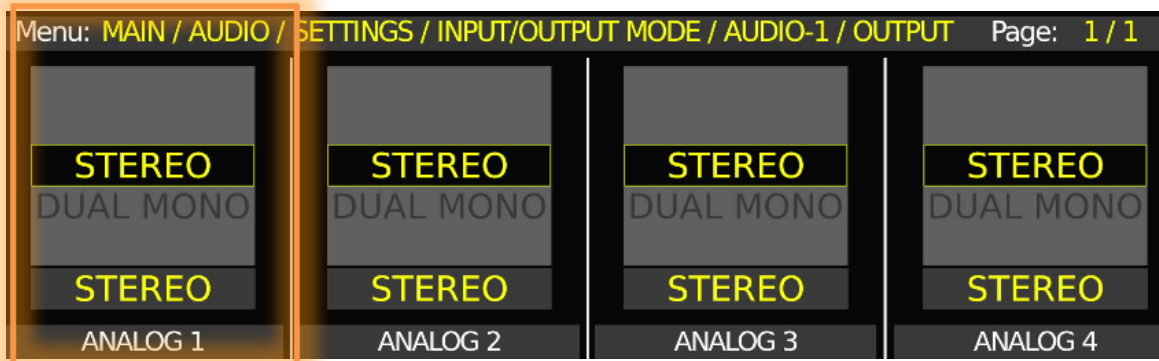
AUDIO-1 OUTPUT configures the four analogue outputs belonging to the first Audio I/O Board.

The page includes:

- **ANALOG 1**
- **ANALOG 2**
- **ANALOG 3**
- **ANALOG 4**

Each output can be configured independently.

ANALOG 1:



ANALOG 1 configures the first balanced XLR analogue output pair of AUDIO-1.

The selectable options are:

- **STEREO:** ANALOG 1 operates as one Left/Right balanced output. This configuration is normally used for a main programme feed, external processor, recorder or other stereo destination.
- **DUAL MONO:** The Left and Right connections become two independent mono outputs. If you need a Telco Output, switch it to mono and after assign to it the desired Telco output.

ANALOG 2:



ANALOG 2 configures the second balanced XLR analogue output pair of AUDIO-1.

The selectable options are:

- **STEREO**
- **DUAL MONO**

ANALOG 2 uses the same operating logic as ANALOG 1. **Important:** ANALOG 2 is not reserved for any specific function. It can remain a normal stereo or dual-mono output depending on system requirements. If you need a Telco Output, switch it to mono and after assign to it the desired Telco output.

ANALOG 3:

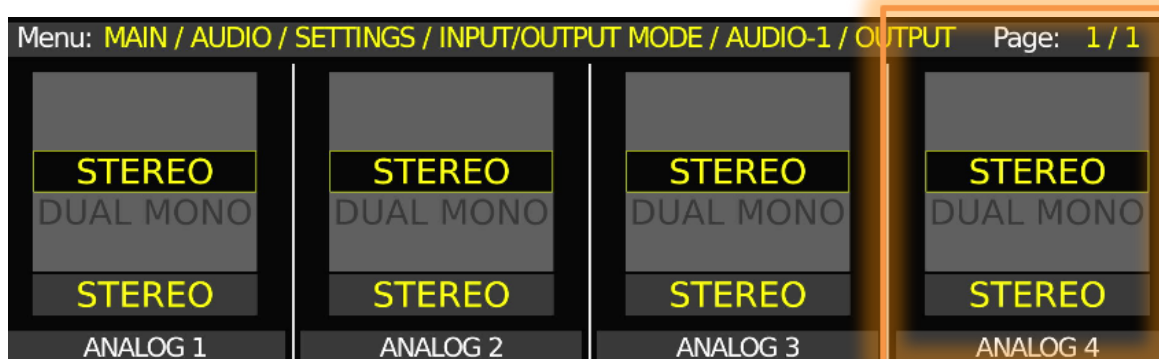
ANALOG 3 configures the first balanced RJ45 analogue output pair of AUDIO-1.

The selectable options are:

- **STEREO**
- **DUAL MONO**

The audio behaviour is identical to ANALOG 1 and ANALOG 2. The principal difference is the physical connector. When connecting ANALOG 3:

- Use the correct RJ45 audio pinout
- Use a compatible breakout adapter when connecting XLR equipment

ANALOG 4:

ANALOG 4 configures the first balanced RJ45 analogue output pair of AUDIO-1.

The selectable options are:

- **STEREO**
- **DUAL MONO**

The audio behaviour is identical to ANALOG 1 and ANALOG 2. The principal difference is the physical connector. When connecting ANALOG 4:

- Use the correct RJ45 audio pinout
- Use a compatible breakout adapter when connecting XLR equipment

-AUDIO 2 OUTPUT:

AUDIO-2 OUTPUT configures the four analogue outputs belonging to the optional second Audio I/O Board. The page contains the corresponding output positions:

- **ANALOG 5**
- **ANALOG 6**
- **ANALOG 7**
- **ANALOG 8**

These outputs use the same configuration logic as AUDIO-1:

- ANALOG 4 and ANALOG 5 use balanced XLR connections
- ANALOG 7 and ANALOG 8 use balanced RJ45 connections
- Each output can be set independently to STEREO or DUAL MONO

The second board therefore adds four more balanced analogue output pairs without changing the configuration of the first board.



ATTENTION: Detailed information about all configurable options for **ANALOG 5–8** is provided in the previous **AUDIO 1 OUTPUT** subsection within the **AUDIO 1–2 OUTPUT** section.

DANTE (INPUT/OUTPUT)



-DANTE INPUT:

The **DANTE INPUT** page, located under **INPUT/OUTPUT MODE**, defines how the incoming Dante Audio-over-IP channels are presented to the internal routing system of the OXYGEN 3000 Plus. It can be accessed through: **MAIN → AUDIO → SETTINGS → INPUT/OUTPUT MODE → DANTE → INPUT**

The page includes eight consecutive Dante input pairs:

- **DANTE 1-2 MODE**

- DANTE 3-4 MODE
- DANTE 5-6 MODE
- DANTE 7-8 MODE
- DANTE 9-10 MODE
- DANTE 11-12 MODE
- DANTE 13-14 MODE
- DANTE 15-16 MODE

Each parameter controls a pair of incoming Dante network channels and determines whether that pair is presented to the console as:

- One stereo source
- Two independent mono sources
- Two independent TELCO sources

The optional Dante interface provides **16 inputs and 16 outputs** or **32 inputs and 32 outputs**, depending on the installed version, and supports Dante, AES67 and SMPTE ST 2110-30 audio transport.

Important: DANTE INPUT MODE defines only how the network channels are interpreted inside the console. It does not create Dante subscriptions, select a transmitting device or modify network, clock and latency settings.

DANTE 1-2 MODE:



DANTE 1-2 MODE defines how Dante input channels 1 and 2 are used.

The selectable options are:

- STEREO: Channels 1 and 2 operate as one stereo source: Channel 1 → Left. Channel 2 → Right.
- 2 MONO: Channels 1 and 2 become two independent mono sources.
- 2 TELCO: Channels 1 and 2 become two independent TELCO input paths.

DANTE 3-4 MODE:



DANTE 3-4 MODE configures Dante input channels 3 and 4. The selectable options are:

- **STEREO:** Channels 1 and 2 operate as one stereo source: Channel 1 → Left. Channel 2 → Right.
- **2 MONO:** Channels 1 and 2 become two independent mono sources.
- **2 TELCO:** Channels 1 and 2 become two independent TELCO input paths.

DANTE 5-6 MODE:



DANTE 5-6 MODE configures Dante input channels 5 and 6. The selectable options are:

- **STEREO:** Channels 1 and 2 operate as one stereo source: Channel 1 → Left. Channel 2 → Right.
- **2 MONO:** Channels 1 and 2 become two independent mono sources.
- **2 TELCO:** Channels 1 and 2 become two independent TELCO input paths.

DANTE 7-8 MODE:



DANTE 7-8 MODE configures Dante input channels 7 and 8. The selectable options are:

- **STEREO:** Channels 1 and 2 operate as one stereo source: Channel 1 → Left. Channel 2 → Right.
- **2 MONO:** Channels 1 and 2 become two independent mono sources.
- **2 TELCO:** Channels 1 and 2 become two independent TELCO input paths.

DANTE 9-10 MODE:



DANTE 9-10 MODE configures Dante input channels 9 and 10. The selectable options are:

- **STEREO:** Channels 1 and 2 operate as one stereo source: Channel 1 → Left. Channel 2 → Right.
- **2 MONO:** Channels 1 and 2 become two independent mono sources.
- **2 TELCO:** Channels 1 and 2 become two independent TELCO input paths.

DANTE 11-12 MODE:



DANTE 11-12 MODE configures Dante input channels 11 and 12. The selectable options are:

- **STEREO:** Channels 1 and 2 operate as one stereo source: Channel 1 → Left. Channel 2 → Right.
- **2 MONO:** Channels 1 and 2 become two independent mono sources.
- **2 TELCO:** Channels 1 and 2 become two independent TELCO input paths.

DANTE 13-14 MODE:



DANTE 13-14 MODE configures Dante input channels 13 and 14. The selectable options are:

- **STEREO:** Channels 1 and 2 operate as one stereo source: Channel 1 → Left. Channel 2 → Right.
- **2 MONO:** Channels 1 and 2 become two independent mono sources.
- **2 TELCO:** Channels 1 and 2 become two independent TELCO input paths.

DANTE 15-16 MODE:



DANTE 15-16 MODE configures Dante input channels 15 and 16. The selectable options are:

- **STEREO:** Channels 1 and 2 operate as one stereo source: Channel 1 → Left. Channel 2 → Right.
- **2 MONO:** Channels 1 and 2 become two independent mono sources.
- **2 TELCO:** Channels 1 and 2 become two independent TELCO input paths.

Important: On a console equipped with the 32-channel Dante option, additional input pairs may be displayed on another page and use the same operating logic.

Hardware note: if the optional Dante interface is not installed or detected, DANTE INPUT parameters are unavailable and the related sources cannot be used.

-DANTE OUTPUT

The **DANTE OUTPUT** page, located under **INPUT/OUTPUT MODE**, defines how the outgoing Dante Audio-over-IP channels are presented to the internal routing system and transmitted from the OXYGEN 3000 Plus to the Dante network. It can be accessed through: **MAIN → AUDIO → SETTINGS → INPUT/OUTPUT MODE → DANTE → OUTPUT**

The page includes eight consecutive Dante output pairs:

- **DANTE 1-2 MODE**
- **DANTE 3-4 MODE**
- **DANTE 5-6 MODE**
- **DANTE 7-8 MODE**
- **DANTE 9-10 MODE**
- **DANTE 11-12 MODE**
- **DANTE 13-14 MODE**
- **DANTE 15-16 MODE**

Each parameter determines whether the corresponding pair of Dante network channels operates as:

- One stereo Left/Right output
- Two independent mono outputs
- Two independent TELCO outputs

The optional Dante interface provides **16 inputs and 16 outputs** or **32 inputs and 32 outputs**, depending on the installed version. The Dante channels can be configured for mono, stereo or TELCO operation.

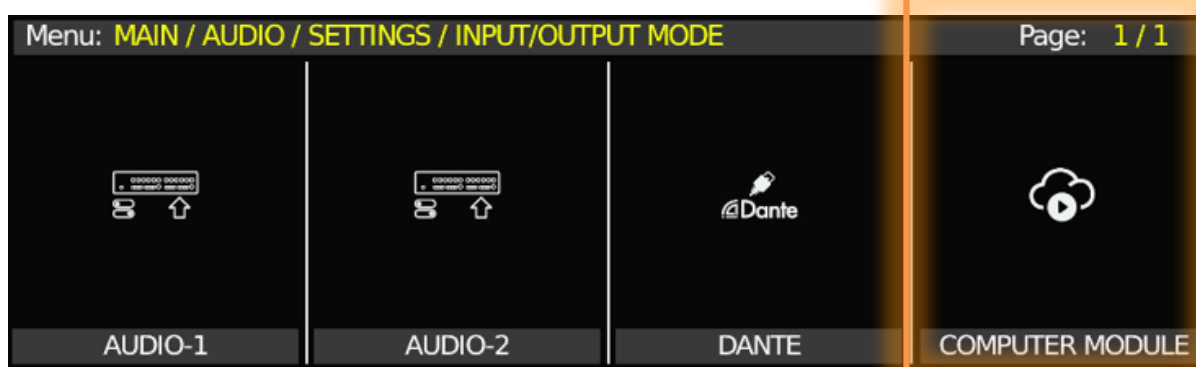
Important: DANTE OUTPUT MODE defines the structure of the network channels. It does not select which console signal is transmitted and does not create subscriptions with receiving Dante devices.

Default-state note: Dante outputs are normally initialised in **STEREO** mode. Change them to 2 MONO or 2 TELCO only when the network routing requires separate channels.

ATTENTION: Detailed information about all configurable options for **DANTE 1-16 MODE** is provided in the previous **DANTE INPUT** subsection within the **DANTE (INPUT/OUTPUT)** section.

COMPUTER MODULE

The **COMPUTER MODULE** page, located under **INPUT/OUTPUT MODE**, determines whether the console's computer-based audio resources are used for network audio streaming or for bidirectional VoIP communication. It can be accessed through: **MAIN → AUDIO → SETTINGS → INPUT/OUTPUT MODE → COMPUTER MODULE**



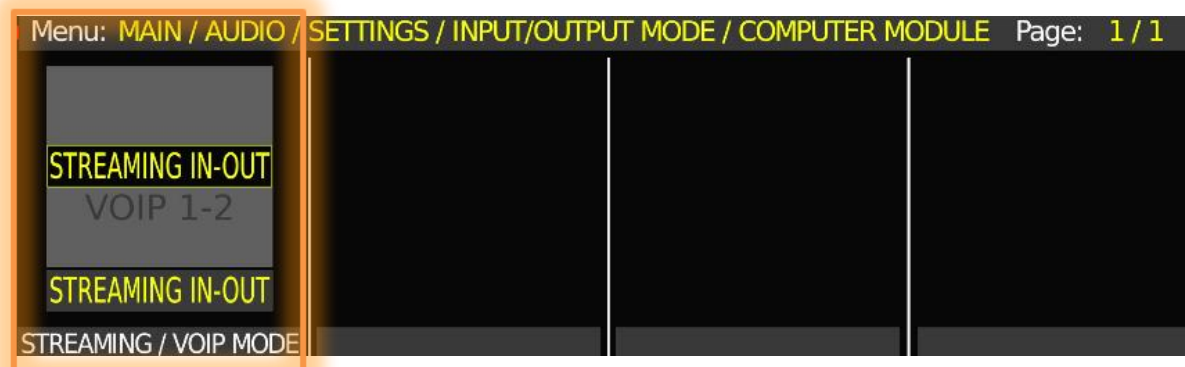
The page includes:

- **STREAMING/VOIP MODE**

Unlike the AUDIO-1, AUDIO-2 and DANTE mode parameters, STREAMING/VOIP MODE does not divide an audio pair into stereo or mono channels. It selects the principal application assigned to the Computer Module. The selectable options are:

- **STREAMING**
- **VOIP**

Important: STREAMING and VOIP use alternative logical resources. Changing the mode may remove the currently available streaming or VoIP sources from the console's routing menus.



ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

STREAMING IN-OUT:

STREAMING assigns the Computer Module to network audio encoding and decoding. In this mode, the OXYGEN 3000 Plus can:

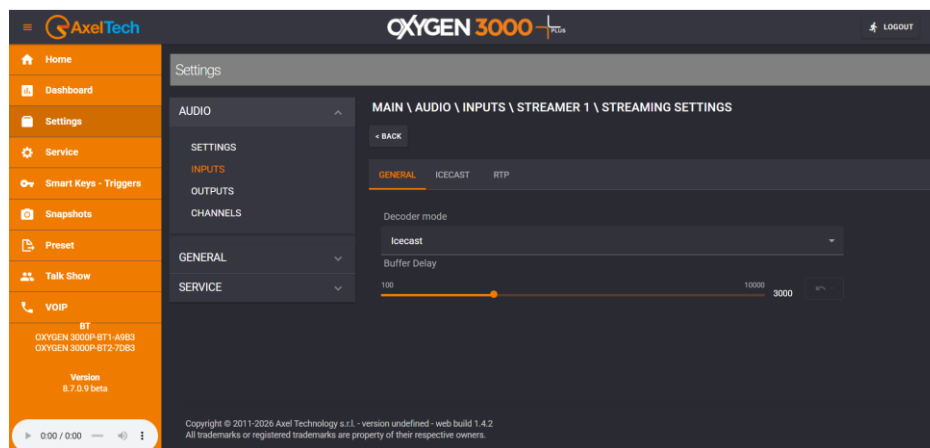
- Receive an audio stream from a network or the Internet
- Decode the received stream and make it available as an input source
- Encode an internal console signal
- Transmit the encoded signal over a network
- Operate with unicast or multicast audio connections
- Integrate with supported streaming servers and network-audio services

The OXYGEN 3000 Plus combines an audio **decoder** for incoming streams with an **encoder** for outgoing streams. It has one streaming input and one streaming output and supports common audio formats such as AAC, Linear PCM, MP2, MP3, Ogg Vorbis and Opus.

Streaming input — DECODER:

The decoder receives an audio stream and converts it into a source that can be used by the console. The streaming input can normally be configured through the console's web interface under the STREAMER input settings. The related path may be displayed as:

WEB INTERFACE → MAIN → AUDIO → INPUTS → STREAMER 1 → STREAMING SETTINGS



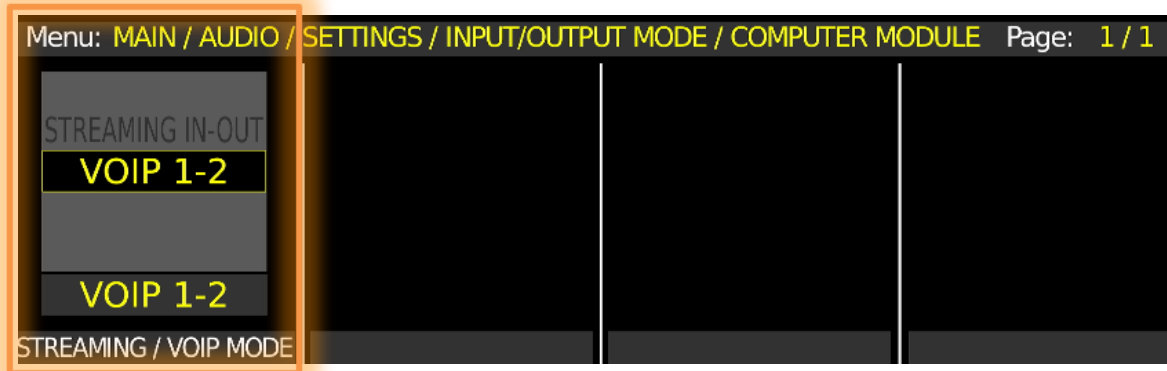
Important: selecting STREAMING does not automatically connect to a remote stream. The stream address, protocol, codec, port and related network parameters must be configured separately.

Streaming output — ENCODER:

The encoder receives an internal console signal, compresses it according to the selected codec and transmits it over the network. The streaming output can normally be configured through the corresponding STREAMER output or web-interface page.

Important: the STREAMING/VOIP MODE parameter enables the streaming resource but does not select the encoder source. The audio sent to the encoder must still be assigned through the appropriate OUTPUTS menu.

VOIP 1-2:



VOIP assigns the Computer Module to bidirectional Voice over IP communication. In this mode, the module exchanges audio with compatible computer-based calling, conferencing or communication software. Depending on the installed software and firmware, the console may provide logical sources such as:

- **VOIP 1**
- **VOIP 2**

When supported by the installed Computer Module, VOIP mode may create two independent communication sources. VOIP 1: The first software-based call path. It can be assigned to a channel and configured independently from VOIP 2. VOIP 2: The second software-based call path.

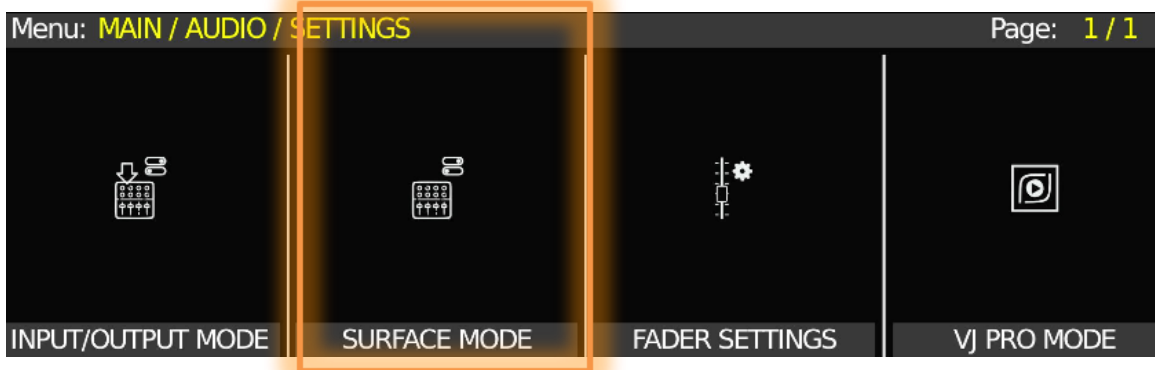
VOIP 1 and VOIP 2 may operate simultaneously when supported by the installed computer hardware and software.

Warning: changing STREAMING/VOIP MODE while a stream or call is On Air may immediately interrupt the audio connection. Perform the change only when the Computer Module is out of service.

1.2 SURFACE MODE

This function is useful to limit the rights to change settings using the surface.

The **SURFACE MODE** menu, located under **SETTINGS**, defines how selected controls on the OXYGEN 3000 Plus surface can be used by the operator. It can be accessed through: **MAIN → AUDIO → SETTINGS → SURFACE MODE**



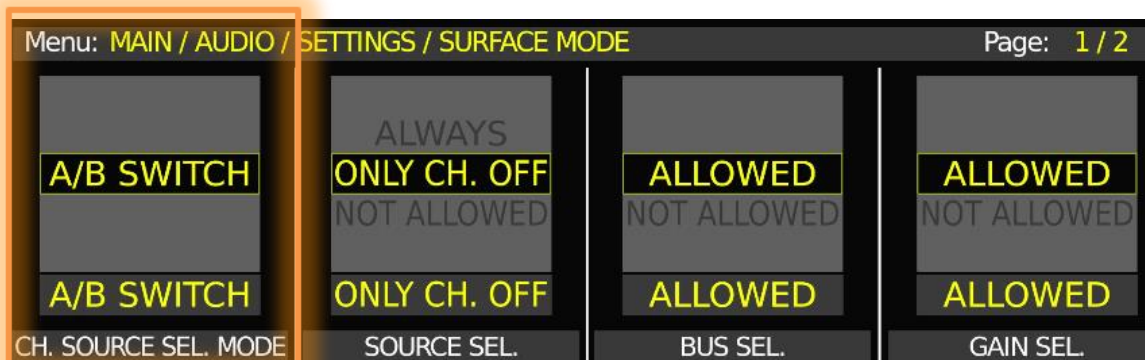
The SURFACE MODE section includes:

- **CH. SOURCE SEL. MODE**
- **SOURCE SEL.**
- **BUS SEL.**
- **GAIN SEL.**
- **CONFIGURATION MENU**
- **VIRTUAL CHANNELS**
- **LEAVE VIRTUAL CHANNELS ON**

These parameters are particularly useful when the console must be adapted to different operating environments, such as: Fully configurable production studios, On Air studios with restricted controls, Shared control rooms, Training installations.

Important: SURFACE MODE controls the availability and behaviour of selected operator functions. It does not directly change audio routing, source GAIN, bus assignments or virtual-channel configuration until the corresponding control is operated.

CH. SOURCE SEL. MODE:



CH. SOURCE SEL. MODE defines how the operator selects the audio source associated with a physical channel.

The selectable option is:

- **A/B SWITCH:** This parameter defines the selection method. The separate **SOURCE SEL.** parameter determines whether and when that method can be used.

A/B SWITCH limits the normal source-selection operation to the two sources already assigned to:

- **CH-A**
- **CH-B**

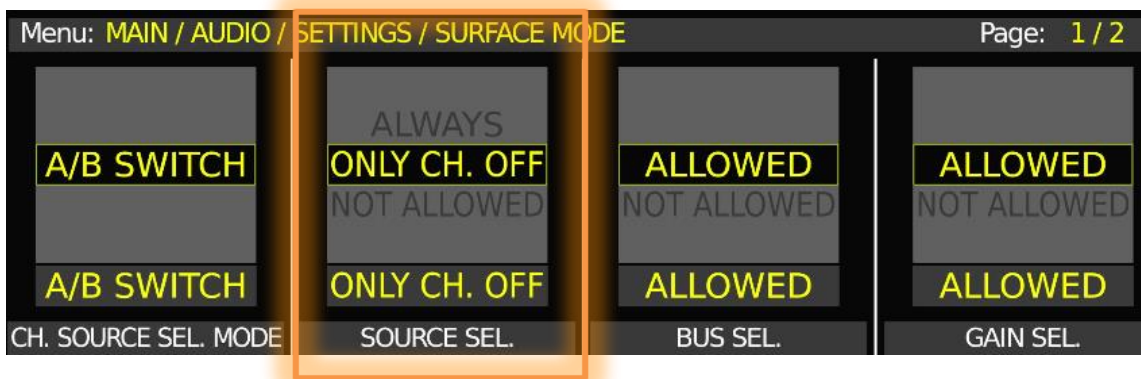
Pressing the channel's **GAIN** encoder switches between the A and B source positions, subject to the restrictions defined by **SOURCE SEL.** In this mode:

- The operator can quickly alternate between two predefined sources
- The complete input-source list is not opened during normal operation
- The CH-A and CH-B assignments remain configured in advance

A/B SWITCH is useful when one fader must provide immediate access to two known sources

This settings can not be modified.

SOURCE SEL.



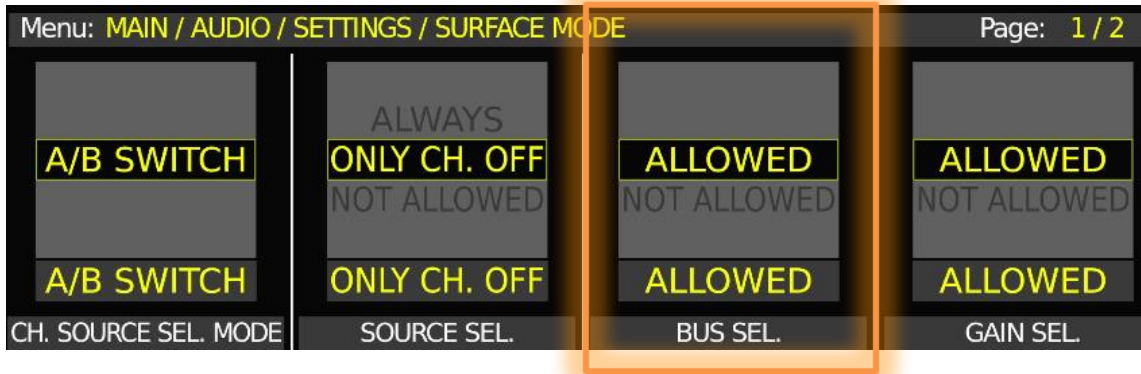
SOURCE SEL. determines when the operator is permitted to change the source assigned to a channel.

The selectable options are:

- **ALWAYS:** When **SOURCE SEL.** is set to **ALWAYS**, the operator can change the channel source at any time. With **A/B SWITCH**, pressing the **GAIN** encoder can switch between CH-A and CH-B at any time. This mode provides maximum flexibility but should be used carefully.
- **ONLY CH.OFF:** When **SOURCE SEL.** is set to **ONLY CH.OFF**, the source can be changed only while the corresponding channel is OFF. If the channel is ON, the source-selection command is blocked until the channel is switched OFF. **ONLY CH.OFF** is normally suitable for live-broadcast environments where the operator must retain source-selection access without being able to change an active channel accidentally.
- **NOT ALLOWED:** When **SOURCE SEL.** is set to **NOT ALLOWED**, the operator cannot change the channel source from the normal surface controls. **Important:** **NOT ALLOWED** does not remove or mute the source. It only locks the source-selection operation.

The restriction applies to the method selected under **CH. SOURCE SEL. MODE**.

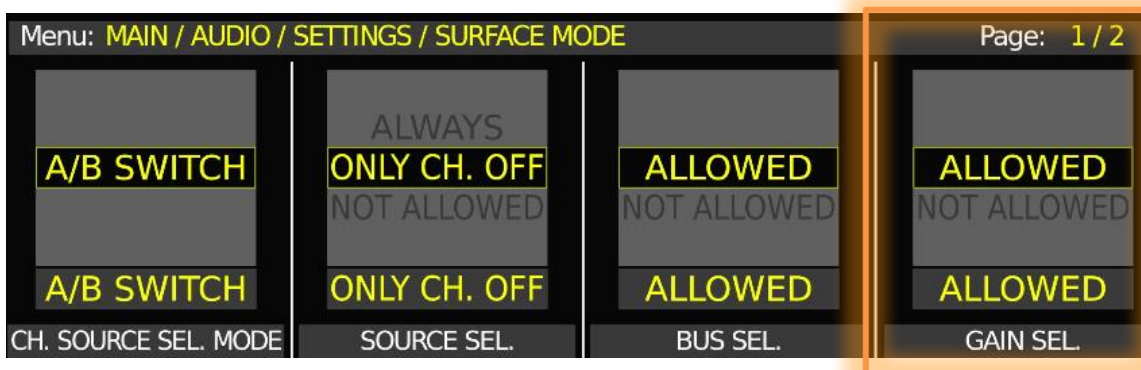
BUS SEL.:



BUS SEL. determines whether the operator can change the buses to which a channel is assigned. The selectable options are:

- **ALLOWED:** When BUS SEL. is set to **ALLOWED**, the operator can use the channel's bus-selection controls normally. This is suitable for production studios where routing must be modified during normal operation. **Important:** allowing bus selection does not bypass other channel restrictions. AUX availability, telephone routing, direct-output configuration and source-specific limitations continue to apply.
- **NOT ALLOWED:** When BUS SEL. is set to **NOT ALLOWED**, bus-assignment changes from the normal surface controls are blocked. The existing routing remains active, but the operator cannot normally add or remove the channel from the available buses. The bus buttons or display indicators may continue to show the current assignments even though changes are not permitted. **Important:** NOT ALLOWED locks the operator control; it does not automatically remove the channel from its currently assigned buses.

GAIN SEL.:

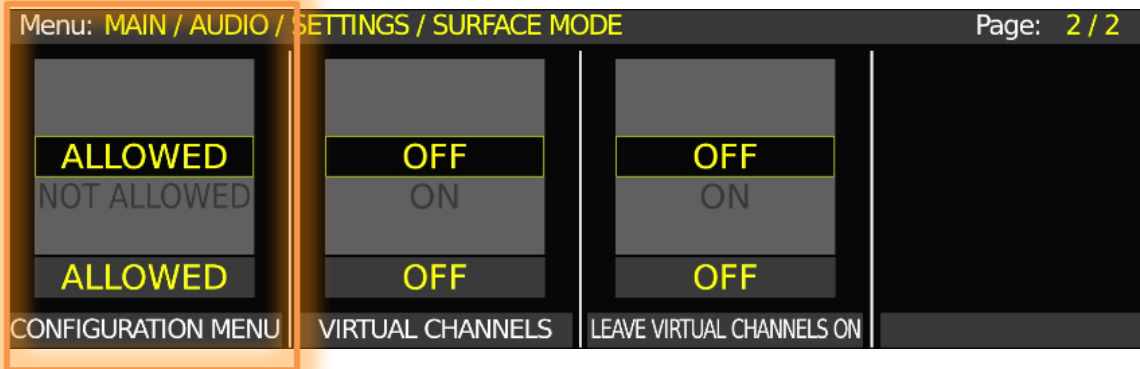


GAIN SEL. determines whether the operator can adjust the selected source's GAIN from the channel surface. The selectable options are:

- **ALLOWED:** When GAIN SEL. is set to **ALLOWED**, the operator can use the GAIN encoder to adjust the input level within the range available for the selected source. GAIN adjustment should be performed while observing the channel meter and, where possible, listening through PFL.
- **NOT ALLOWED:** When GAIN SEL. is set to **NOT ALLOWED**, source-GAIN adjustment from the normal surface control is blocked. The stored GAIN value remains applied.

Important: GAIN SEL. and SOURCE SEL. are separate permissions. Depending on the selected operating mode, pressing the GAIN encoder may still perform an authorised source-selection function even when rotation for GAIN adjustment is locked.

CONFIGURATION MENU:

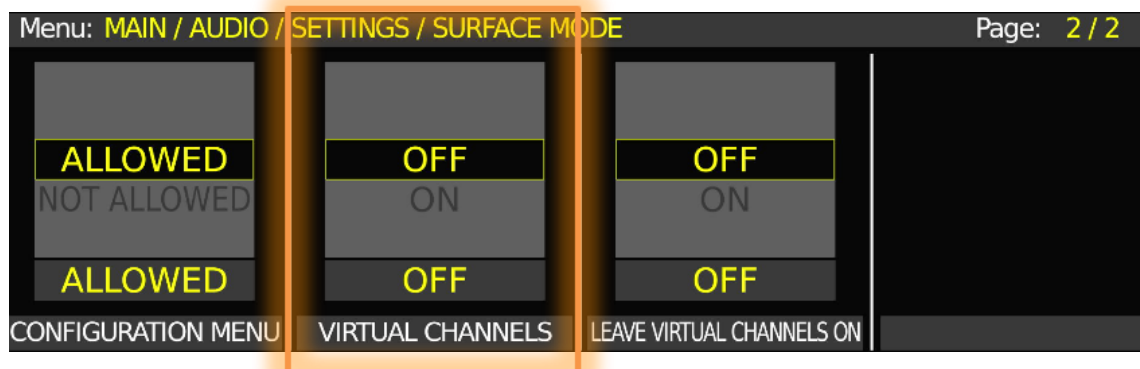


CONFIGURATION MENU determines whether the operator can enter the console's configuration pages from the surface. The selectable options are:

- **ALLOWED:** When CONFIGURATION MENU is set to **ALLOWED**, configuration pages can be opened normally, subject to any additional password or access restrictions provided by the system.
- **NOT ALLOWED:** When CONFIGURATION MENU is set to **NOT ALLOWED**, access to the protected configuration area from the normal surface workflow is blocked. However, the operator cannot normally open the configuration pages controlled by this permission. This mode is useful for preventing accidental modification of system setting

Important: CONFIGURATION MENU restrictions are independent from SOURCE SEL., BUS SEL. and GAIN SEL. Blocking the configuration menu does not automatically lock the normal operational controls; each permission must be configured separately.

VIRTUAL CHANNELS:



VIRTUAL CHANNELS determines whether the operator can access and operate the virtual-channel layer from the console surface. The selectable options are:

- **ON:** When VIRTUAL CHANNELS is set to **ON**, the operator can enter the virtual-channel layer and use the configured virtual channels. The separate global virtual-channel function must also be enabled, normally through the corresponding CHANNELS configuration page.

- **OFF:** When VIRTUAL CHANNELS is set to **OFF**, the operator cannot access the virtual-channel layer from the normal surface controls. Configured virtual channels are not deleted. Depending on their current state and on **LEAVE VIRTUAL CHANNELS ON**, they may: Remain active in the routing system, Continue to pass audio, Remain controllable from an authorised remote interface, Be unavailable only from the local surface

Virtual channels are additional logical channels that are not permanently associated with a dedicated physical fader. When enabled and configured, they can provide control of extra sources through the console interface or OXYGEN REMOTER. The OXYGEN 3000 Plus can provide up to eight virtual channels, depending on firmware and configuration. Their source assignments and operating parameters are managed separately from the SURFACE MODE permission.

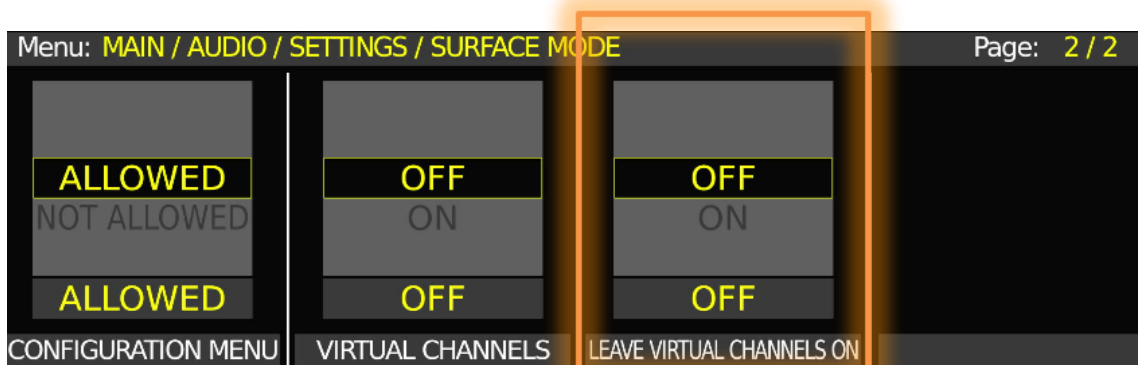
Important: blocking access to virtual channels does not necessarily switch them OFF. Always verify their current ON/OFF and routing states before restricting access.

LEAVE VIRTUAL CHANNELS ON:

LEAVE VIRTUAL CHANNELS ON defines what happens to active virtual channels when the operator exits the virtual-channel layer and returns to the standard surface view. The selectable options are:

- **ON:** When LEAVE VIRTUAL CHANNELS ON is set to **ON**, leaving the virtual-channel layer does not automatically switch active virtual channels OFF. **Warning:** a virtual channel can remain On Air even when it is no longer displayed on the physical surface. The operator must verify its state before leaving the virtual-channel layer.
- **OFF:** When LEAVE VIRTUAL CHANNELS ON is set to **OFF**, the console does not leave active virtual channels ON when the operator exits the virtual-channel layer. The affected virtual channels are normally switched OFF or prevented from remaining active, according to the installed firmware's channel logic.

This parameter is important because a virtual channel may continue passing audio even when it is no longer visible on the physical surface. **Important:** before using this option, verify whether the virtual channels carry automation, streaming or other continuous services. Automatically switching them OFF may interrupt programme audio.



1.3 FADER SETTINGS

The **FADER SETTINGS** menu, located under **SETTINGS**, defines the operating behaviour of the OXYGEN 3000 Plus channel faders. It can be accessed through: **MAIN → AUDIO → SETTINGS → FADER SETTINGS**



The menu includes:

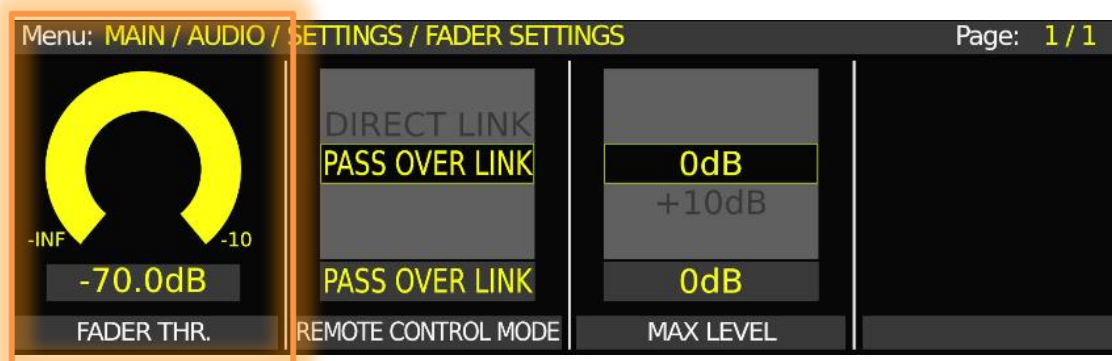
- **FADER THR.**
- **REMOTE CONTROL MODE**
- **MAX LEVEL**

These parameters determine:

- The fader position at which a channel is considered ON or OFF
- How control is transferred between a physical fader and the corresponding virtual fader in OXYGEN REMOTER
- The maximum gain available at the top of the fader travel

Important: FADER SETTINGS does not adjust the input PREAMP or source GAIN. It controls the channel fader's operating threshold, remote-control relationship and maximum level.

FADER TRH.:



FADER THRESHOLD, defines the channel-fader level at which the console changes between the logical OFF and ON conditions. FADER THR. is particularly relevant when the selected source's **ON MODULE** configuration includes fader-controlled activation, such as: **BY FADER, FADER & BUTTON, BY FADER & COUGH BUTTON** Range: **-70.0 dB. to -10.0 dB. Adjustment step: 1.0 dB.** The threshold divides the fader travel into two operating areas:

- **Fader above the threshold:** the channel is considered ON.
- **Fader below the threshold:** the channel is considered OFF.

For example, if FADER THR. is set to **−50.0 dB**:

- Raising the fader above −50 dB activates the channel.
- Lowering the fader below −50 dB deactivates the channel.

Important: the threshold defines a channel-status transition, not only an audio attenuation value. When the ON/OFF status changes, other functions linked to that status may also be activated or deactivated.

Warning: if FADER THR. is set close to −10 dB, the source may become audible suddenly at a high level when the threshold is crossed. Verify the source GAIN and monitor the result before using a high threshold On Air.

REMOTE CONTROL MODE:



REMOTE CONTROL MODE defines the relationship between each physical channel fader and its corresponding virtual fader in **OXYGEN REMOTER**. The parameter may be abbreviated as **REMOTE CTRL** on the console display. The selectable options are:

- **DIRECT LINK:** **DIRECT LINK** keeps the physical fader directly connected to the corresponding virtual fader. In this mode: Moving the physical fader immediately moves the virtual fader, The physical fader always has direct control of the channel level, No position-matching procedure is require

Direct-link example

Assume that: The physical fader is positioned at **0 dB** and the virtual fader is moved remotely to **−20 dB**

Because the physical fader cannot physically follow the virtual movement, the two positions are different. When the local operator moves the physical fader, **DIRECT LINK** immediately transfers control to its physical position. The channel level may therefore jump from the remotely selected −20 dB value toward the physical fader value.

Warning: **DIRECT LINK** can produce an abrupt level change when the physical and virtual fader positions are different. Check the virtual level before moving a physical fader that has recently been adjusted remotely.

- **PASS OVER LINK:** **PASS OVER LINK** provides a protected pickup or pass-over handover between the virtual and physical faders. When the virtual fader is moved through **OXYGEN REMOTER**: The corresponding physical fader is temporarily disconnected, Moving the physical fader does not immediately change the virtual level, The remote level remains active.

The required condition is: **Physical Fader Level = Virtual Fader Level**

Once the physical fader passes through or matches the virtual position, the link is restored and subsequent physical movements control the channel normally.

The physical faders of the OXYGEN 3000 Plus are not motorised. Therefore, when a virtual fader is moved remotely, the physical fader cannot automatically move to the same position. REMOTE CONTROL MODE determines what happens when the local and remote positions no longer correspond. **Important:** this parameter controls fader-position handover. It does not enable the network connection, select the remote-control computer or configure OXYGEN REMOTER access.

MAX LEVEL:



MAX LEVEL defines the highest channel level that can be reached by moving a fader to the top of its travel. The selectable options are:

- **+10 dB:** When MAX LEVEL is set to **+10 dB**, the upper part of the fader travel can provide up to 10 dB of gain above the nominal unity position. In this mode: **0 dB** remains the unity-gain position, The fader can be raised above unity, The maximum available level is **+10 dB**, Additional correction is available for sources that are temporarily too low, The complete normal fader range is available.

Warning: positive fader gain reduces the available headroom and can cause bus or output clipping. The input GAIN should normally be aligned so that the source operates close to 0 dB without requiring continuous positive fader gain.

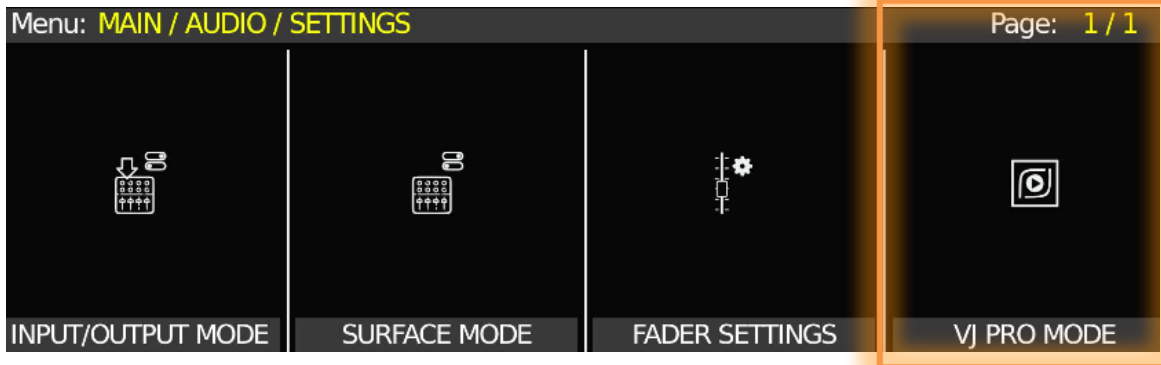
- **0 dB:** When MAX LEVEL is set to **0 dB**, the highest possible fader position corresponds to unity gain. In this mode: The fader can attenuate the source, The fader can reach 0 dB, The fader cannot add positive gain, Operators cannot raise the channel above its nominal level, The possibility of accidental fader boost is reduced

Important: MAX LEVEL = 0 dB does not guarantee that the audio cannot clip. Excessive PREAMP, input GAIN, EQ boost, dynamics makeup gain or bus summing can still produce overload.

This setting applies a maximum limit to the channel-fader range. **Important:** MAX LEVEL controls the channel fader's upper limit. It does not change the source PREAMP, input GAIN or maximum level of the physical audio outputs.

1.4 VJ PRO MODE

The **VJ PRO MODE** menu, located under **SETTINGS**, configures the audio resources used by the OXYGEN 3000 Plus when it is integrated with the Axel Technology **VJ Pro Visual Radio system**. It can be accessed through: **MAIN → AUDIO → SETTINGS → VJ PRO MODE**



The menu includes:

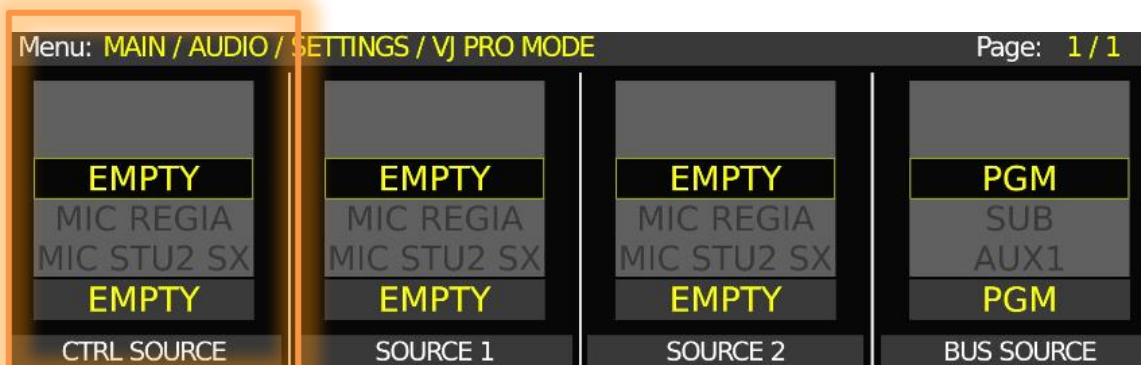
- **CTRL SOURCE**
- **SOURCE 1**
- **SOURCE 2**
- **BUS SOURCE**

VJ PRO MODE links the radio-automation audio source and the two VJ Pro player sources so that their levels can be managed coherently from the OXYGEN 3000 Plus. The VJ Pro system coordinates video clips, cameras, graphics and programme information with the radio automation and mixer activity.

Important: VJ PRO MODE configures the audio-source relationship inside the OXYGEN 3000 Plus. It does not install VJ Pro, configure its video players or establish the network connection with the Visual Radio system.

CTRL SOURCE:

The level of the CTRL SOURCE fader will set automatically the faders levels of the SOURCE1-2.



CTRL SOURCE selects the principal radio-side audio source associated with the VJ Pro control logic.

The selectable options are:

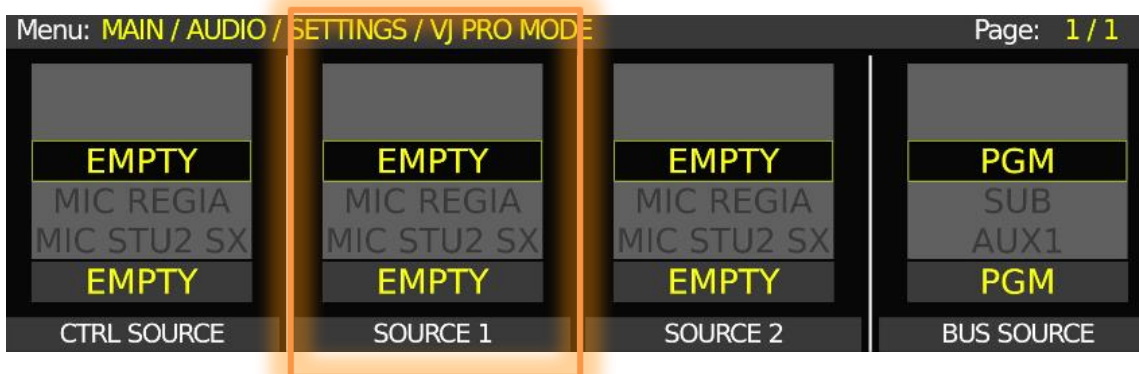
- **EMPTY:** When CTRL SOURCE is set to **EMPTY**, no radio-automation source is assigned to the VJ Pro control group. **Configuration note:** selecting EMPTY does not disable the external VJ Pro software. It only removes the CTRL SOURCE assignment from the console.
- One of the audio input sources currently available in the console

The selected source normally corresponds to the audio output of the radio-automation system.

Important: CTRL SOURCE is an audio source, not a GPI, IP command or video-control input. The communication between DJ Pro and VJ Pro must be configured separately in the associated software.

ATTENTION: if the selected source operates in stereo, verify that its INPUT/OUTPUT MODE is also configured as STEREO and that the Left and Right channels are connected correctly.

SOURCE 1:



SOURCE 1 selects the console input receiving audio from the first VJ Pro multimedia player.

The selectable options are:

- **EMPTY:** When SOURCE 1 is set to **EMPTY**, no console input is assigned to the first VJ Pro player.
- One of the audio input sources currently available in the console

SOURCE 1 normally carries the audio embedded in the video clip corresponding to the radio-automation event currently in progress.

It is suggested to set **SOURCE 1 = LINE 4**. This example assumes that the audio output belonging to the first VJ Pro player is physically connected to the console input presented as LINE 4.

Important: SOURCE 1 does not select a video file inside VJ Pro. It selects the OXYGEN audio input receiving the first player's audio signal.

ATTENTION: if the selected source operates in stereo, verify that its INPUT/OUTPUT MODE is also configured as STEREO and that the Left and Right channels are connected correctly.

SOURCE 2:

SOURCE 2 selects the console input receiving audio from the second VJ Pro multimedia player.

The selectable options are:

- **EMPTY:** When SOURCE 2 is set to **EMPTY**, no console input is associated with the second VJ Pro player.
- One of the audio input sources currently available in the console

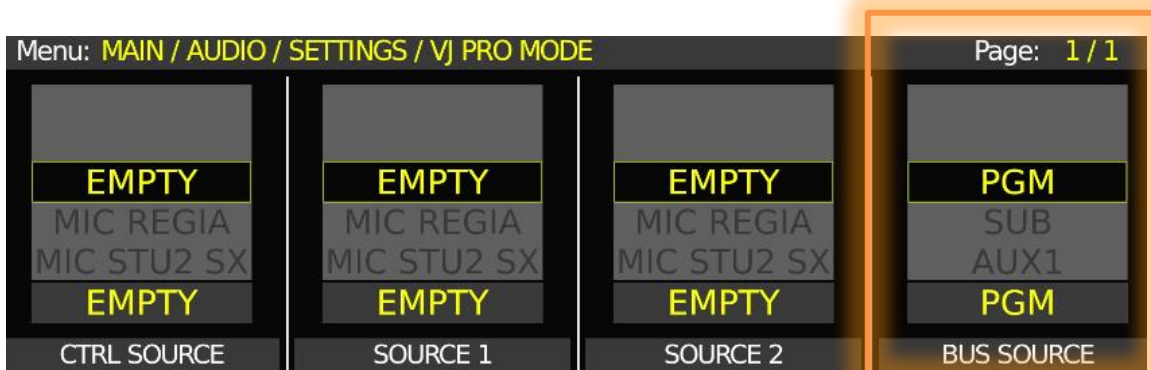
VJ Pro can use two players to prepare transitions between the current clip and the following clip. SOURCE 2 provides the audio path for the second player. The use of two independent players allows VJ Pro to manage smoother transitions and situations in which the duration of the radio audio and corresponding video clip is different.

Warning: verify the active-player sequence before leaving SOURCE 2 empty. VJ Pro may switch between Player 1 and Player 2 even when the clips appear to form one continuous television programme.

ATTENTION: if the selected source operates in stereo, verify that its INPUT/OUTPUT MODE is also configured as STEREO and that the Left and Right channels are connected correctly.

BUS SOURCE:

This selected BUS is where SOURCE 1-2 will be mixed.



BUS SOURCE selects the destination bus used for SOURCE 1 and SOURCE 2. The selectable options are:

- **EMPTY:** When BUS SOURCE is set to **EMPTY**, no automatic destination bus is assigned to SOURCE 1 and SOURCE 2 through VJ PRO MODE.
- **PGM:** When BUS SOURCE is set to **PGM**, SOURCE 1 and SOURCE 2 are routed to the main programme bus. This means that the VJ Pro player audio can become part of the radio On Air signal.

Warning: CTRL SOURCE is already associated with PGM. If a VJ Pro player reproduces the same audio as the radio automation, selecting PGM for BUS SOURCE can duplicate the programme signal, causing increased level, phase interaction or echo.

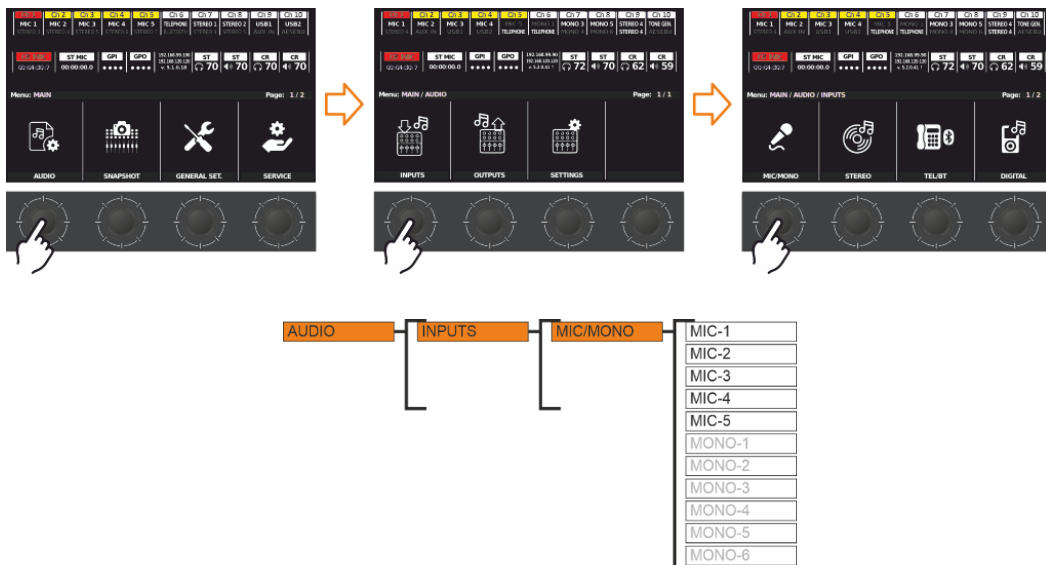
- **SUB:** When BUS SOURCE is set to **SUB**, SOURCE 1 and SOURCE 2 are routed to the SUB bus.
- **AUX1:** When BUS SOURCE is set to **AUX1**, SOURCE 1 and SOURCE 2 are routed to the first auxiliary bus. This is the configuration recommended by Axel Technology for a typical VJ Pro installation.
- **AUX2:** When BUS SOURCE is set to **AUX2**, SOURCE 1 and SOURCE 2 are routed to the second auxiliary bus. AUX2 uses the same operating principle as AUX1.

Configuration note: AUX1 and AUX2 are independent buses. Selecting AUX2 for VJ Pro does not automatically copy sources already assigned only to AUX1.

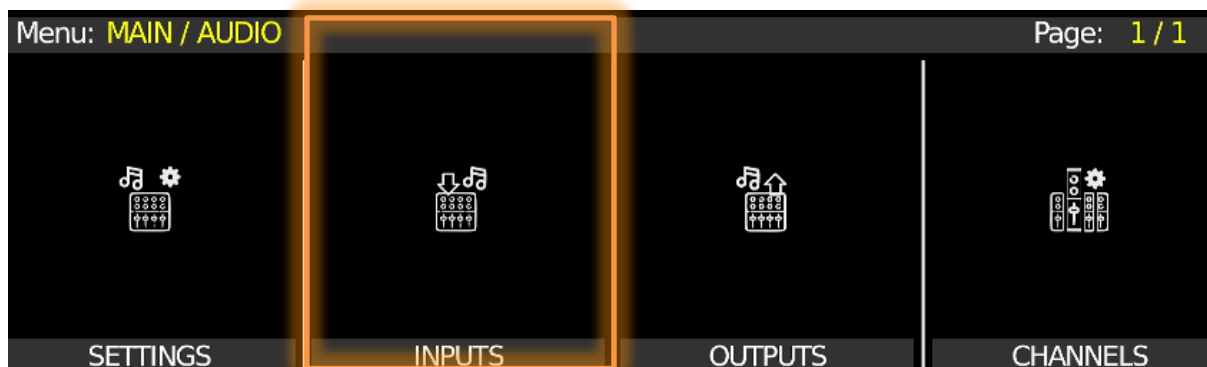
BUS SOURCE applies to the two VJ Pro player sources. It does not normally replace the automatic PGM routing associated with CTRL SOURCE.

2. INPUTS

You can set Input source parameters from the menu **MENU / AUDIO / INPUTS** by PUSH/ROTATE the knobs as in the following steps.

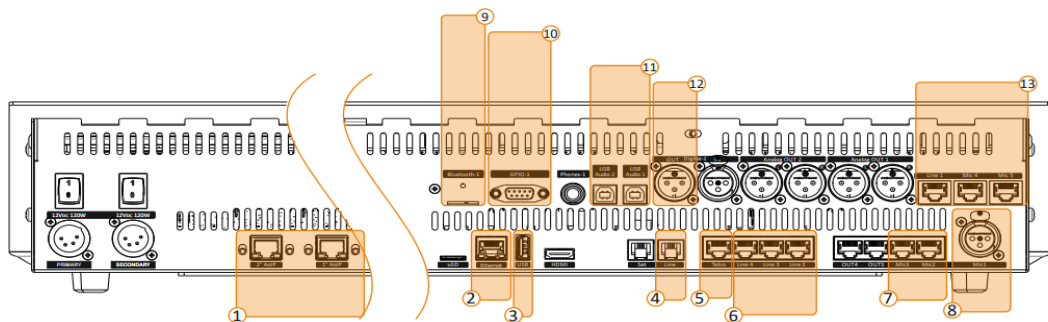


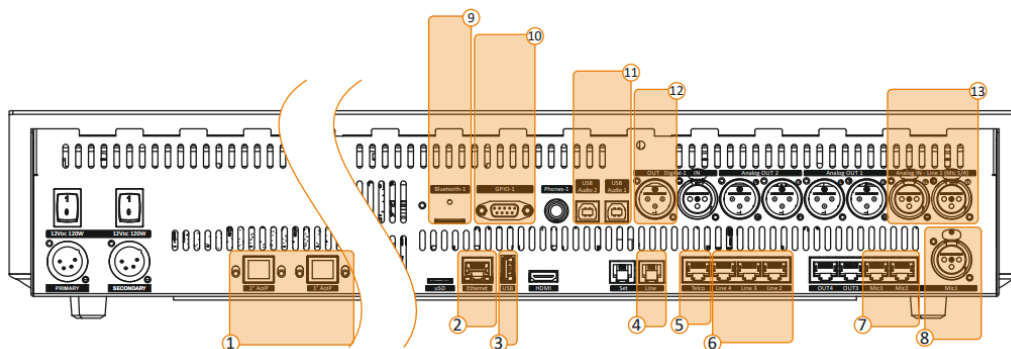
From this level of the menu, you choose which input source you like to customize.



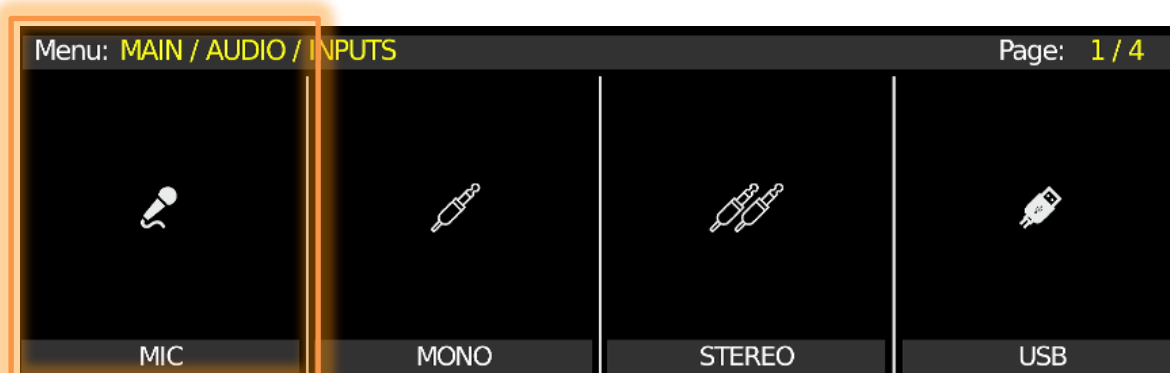
MIC (MIC-1 to MIC-10)

Here we can find all Microphones source parameters. That can manage the microphones' inputs like changing the **NAME**, **GAIN**, **PAL/PAN**, **PREAMP**, **PHASE**, **PHANTOM 48V**, and many other functions we going to explain briefly below.



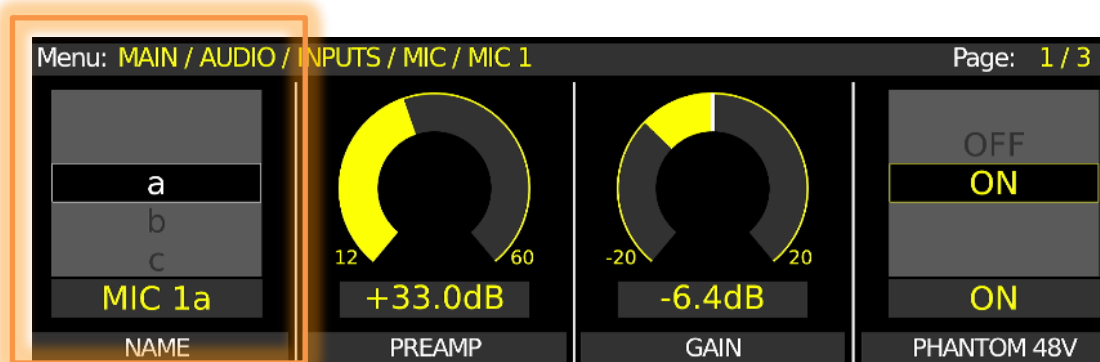


Important: The audio input connections for **MIC 1–5** are located on the **first Audio Board**, at positions **7, 8, and 13**. The connections for **MIC 6–10** are located in the same positions on the **second Audio Board**.

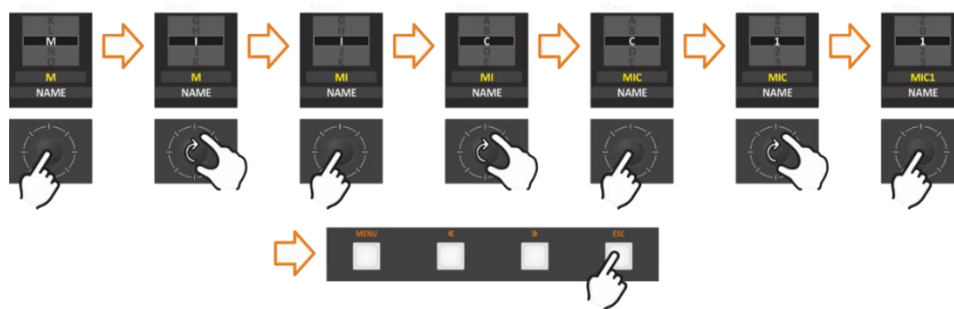


ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

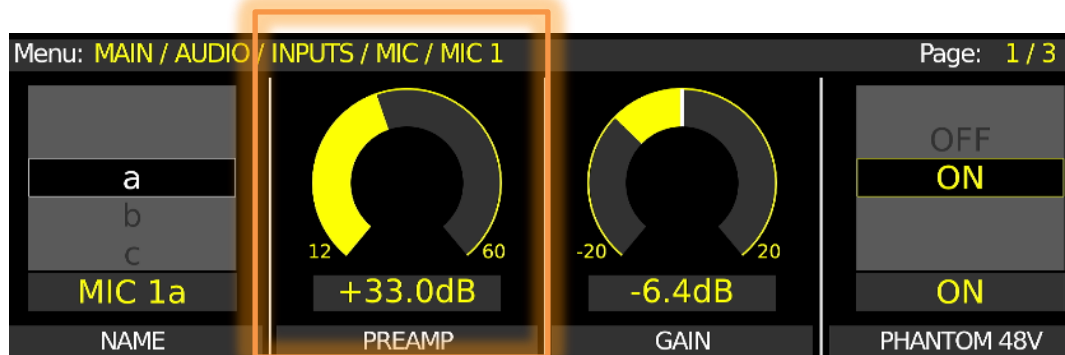
NAME:



This parameter allows the assignment of a LABEL to the various input sources. It is not possible to have a LABEL of more than 8 characters. To change the microphone name and sign it as you like, it's so easy way just rotate and push the knobs. To end editing the LABEL, just push the ESC button. just follow the next steps. The description works the same for all Inputs. (In the example we are going to set the channel name as (MIC1).



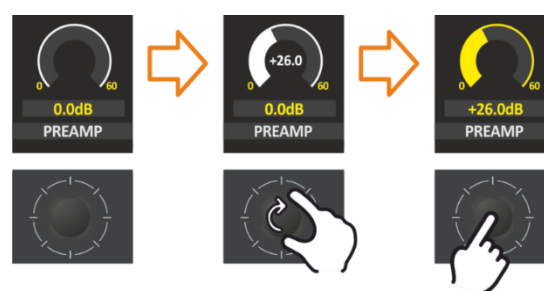
PREAMP:



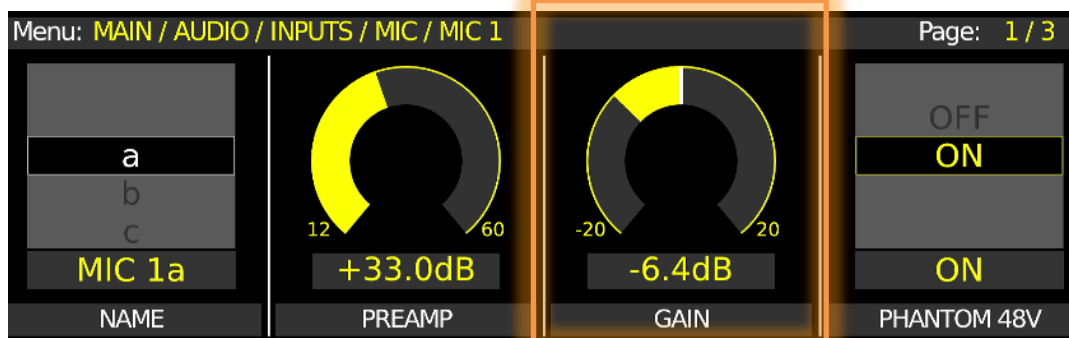
Adjusts the gain applied by the **analogue microphone preamplifier** before the signal is converted into digital audio. It is used to raise the low-level signal produced by a microphone to the correct operating level while maintaining a suitable **signal-to-noise ratio**.

The Preamp should be adjusted so that normal speech produces a healthy input level without activating the channel's clipping indication. Excessive preamplification may cause distortion, while insufficient gain may produce a weak or noisy signal. **Adjustment range: 12.0 dB to 60.0 dB. Step: 3.0 dB.**

N.B.: When you adjust the **PREAMP** parameter, you have a real-time change of the parameter value and a real-time perception of the **PREAMP** change. The Graphics will become white to indicate the parameter change, and they will become yellow again after the confirmation.

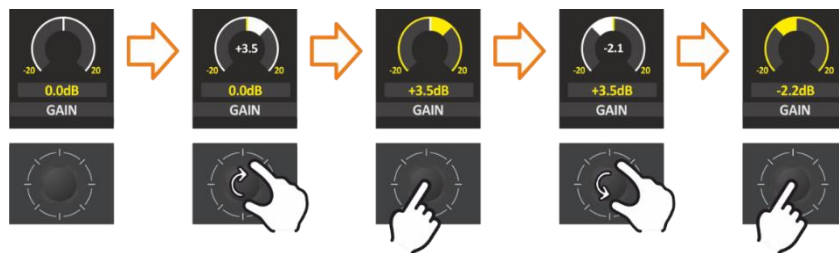


GAIN:

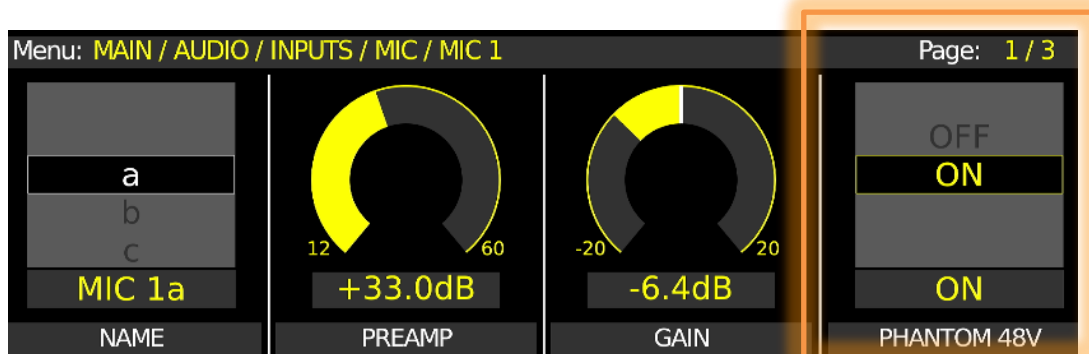


Adjusts the input level of MIC 1 within the mixer's digital processing section. Unlike the Preamp, which controls the analogue amplification stage, Gain provides a precise **digital level trim** after analogue-to-digital conversion. It can be used to make fine corrections after the Preamp has been properly configured. Excessive positive gain may produce digital clipping, while excessive attenuation may reduce the effective output level. **Adjustment range: -20 dB to +20 dB. Step: 0.1 dB.**

N.B.: When the parameter control is activated, in real-time the parameter value is updated, perceiving the increase and decrease of the **GAIN**, the graphics will change from yellow to white until the parameter is confirmed.



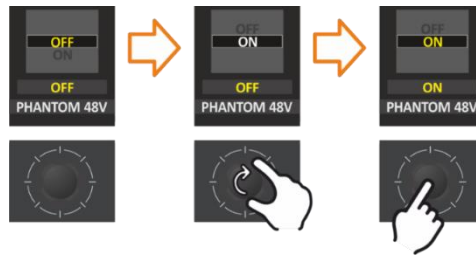
PHANTOM 48V:



Enables or disables the **48 V phantom power supply** on the MIC 1 input. Phantom power is normally required by condenser microphones and certain active microphone devices.

- **OFF:** no phantom voltage is supplied to the microphone input.
- **ON:** 48 V phantom power is supplied through the balanced microphone connection.

Before enabling this function, verify that the connected microphone supports phantom power. It is recommended to **mute or close the channel before switching Phantom 48V on or off**. Extra care is required with passive ribbon microphones or unbalanced equipment.

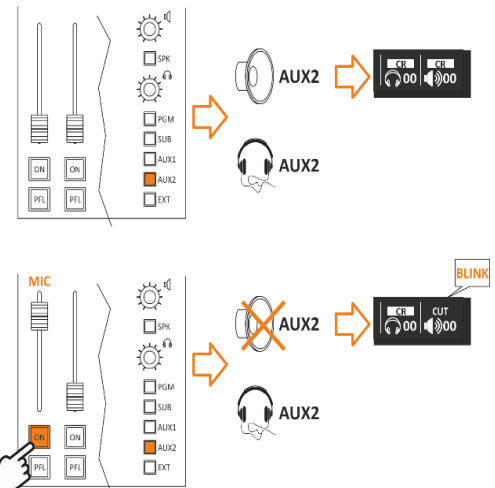


SPK-CUT:

Defines which loudspeaker system is automatically muted when MIC 1 is opened. This prevents the microphone from capturing the loudspeaker signal and generating **acoustic feedback or Larsen effect**.

- **OFF:** opening MIC 1 does not mute any loudspeaker output.
- **CR:** mutes the **Control Room loudspeakers** when MIC 1 is opened.
- **ST:** mutes the **Studio loudspeakers** when MIC 1 is opened.
- **CR+ST:** mutes both the Control Room and Studio loudspeakers.

SPK CUT affects only the configured loudspeaker outputs and does not mute the programme, headphone or recording buses. When you open the associated source with an **SPK-CUT** function, you will see its **MUTE CR** (Control Room) Loudspeakers output and the SPEAKER icon at LCD will start blinking.



ON AIR LIGHT:



Associates MIC 1 with the studio's external **On-Air signalling system**. When the microphone is opened, the console activates the corresponding programmable **GPO command**, which can control an On-Air lamp or another external signalling device.

- **OFF:** MIC 1 does not activate an On-Air Light command.
- **CR:** activates the GPO assigned to the **Control Room On-Air Light**.
- **ST:** activates the GPO assigned to the **Studio On-Air Light**.
- **CR+ST:** activates both the Control Room and Studio On-Air Light commands.

The corresponding GPO outputs must be configured in: **MENU / MAIN / GENERAL SET. / GPIO / GPO-1 BOARD**

PRIVATE MIC:

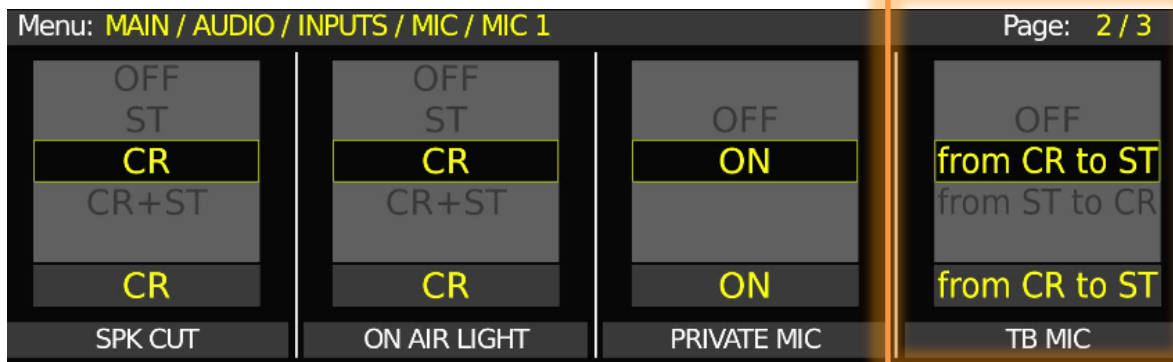


Defines whether MIC 1 can be used for **private off-air communication with a telephone caller**. When this function is enabled, pressing PFL on the corresponding telephone or Telco channel routes MIC 1 to the caller without sending the conversation to the main programme outputs or loudspeakers.

- **OFF:** MIC 1 is not included in the private communication circuit.
- **ON:** MIC 1 can communicate privately with the caller when PFL is activated on the telephone channel.

The Private Mic signal is independent of the channel's **fader position and ON/START status**, allowing communication even when MIC 1 is closed on air.

TB MIC:



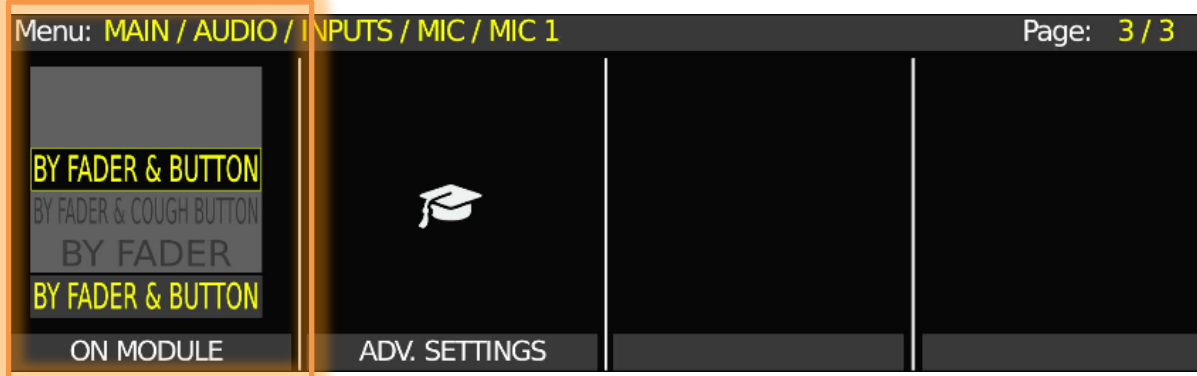
Configures MIC 1 as a **Talkback microphone** and defines the direction of the internal communication. Talkback allows communication between the Control Room operator and presenters or guests without sending the conversation to the programme output.

- **OFF:** MIC 1 is disconnected from the Talkback system.
- **FROM CR TO ST:** MIC 1 is used to communicate **from the Control Room to the Studio**.
- **FROM ST TO CR:** MIC 1 is used to communicate **from the Studio to the Control Room**.

The Talkback signal is routed only to the configured studio or control-room monitoring system.

The Talkback signal is independent of the channel's **fader position and ON/START status**, allowing communication even when MIC 1 is closed on air, it is activated only when a TB button is pressed.

ON MODULE:



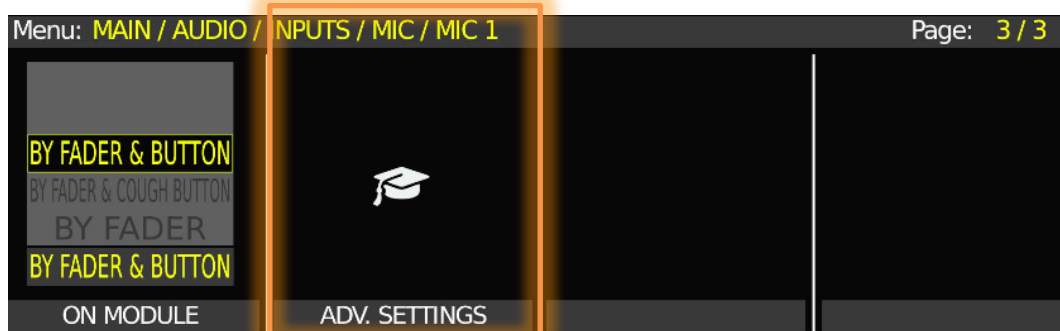
Defines how MIC 1 is switched on or off when connected through an AxelTech **TalkBox module**. The selected mode determines whether the microphone status is controlled by the channel fader, the ON/START or TalkBox button, the Cough button, or a fixed operating condition.

- **FADER & BUTTON:** the microphone status can be controlled through both the channel fader and the ON/START or TalkBox button.
- **BY FADER:** the microphone is switched on when the fader crosses the activation threshold and switched off when it returns below that threshold.
- **BY BUTTON:** the microphone status is controlled by the ON/START button on the console or the corresponding TalkBox button. Moving the fader does not automatically change the ON/OFF status.
- **BY FADER & COUGH BUTTON:** the channel fader controls the normal activation of the microphone, while the **Cough button temporarily mutes the microphone signal** without changing the fader position. The microphone returns to its previous operating state when the Cough button is released. This allows the presenter to suppress coughing or other unwanted sounds without closing the channel.
- **ALWAYS ON:** the microphone remains permanently enabled on the module, independently of the fader and button controls.
- **ALWAYS OFF:** the microphone remains permanently disabled on the module.

The **ALWAYS ON** option should be used carefully because the microphone may remain active independently of the normal console controls.

ADV. SETTINGS:

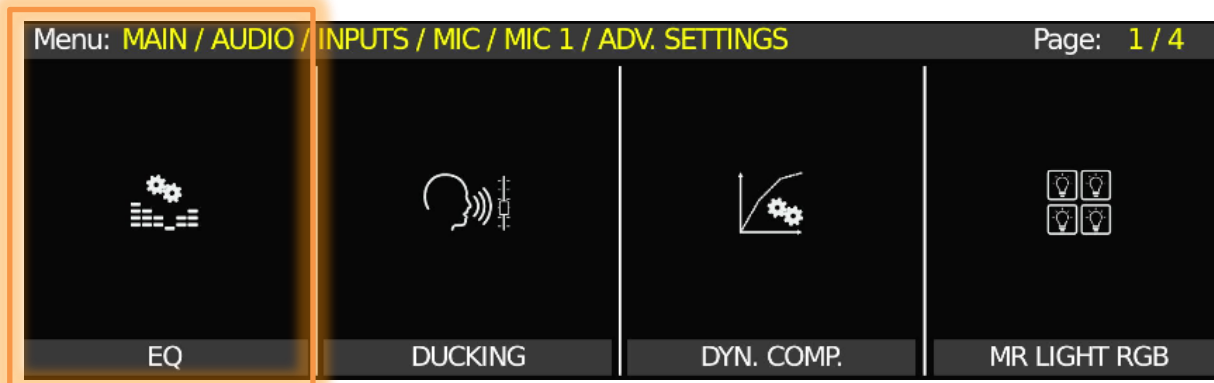
The **ADVANCED SETTINGS** section provides access to the detailed operating parameters of a selected input source or output.



While the main configuration page contains the parameters required for normal operation—such as **NAME**, **GAIN**, **SOURCE** and **ON MODULE**—ADVANCED SETTINGS allows the operator to customise the source's processing, routing, control logic and surface behaviour.

EQ

Parametric EQ: This type of EQ allows you to select a specific frequency to be adjusted with the **FREQUENCY** knob and adjust the bandwidth of the EQ with the **Q** knob. The **Gain** knob allows you to boost or cut the selected frequency.



When adjusting the bandwidth of the EQ, you can choose between a "**Peak**" or "**Shelving**" filter. A Peak filter allows you to adjust the bandwidth of a specific frequency range, while a Shelving filter adjusts the bandwidth of all frequencies above or below a certain point.

These are graphic equalizers that offer a versatile adjustable wide band ranging the spectrum from **31.5 Hz** to **20 kHz**. This equalizer enables the sound to be amplified or reduced in specific frequency bands, which allows for the manipulation of the quality and characteristics of the sound. Furthermore, a maximum boost or cut can be applied to the **LOW-CUT** and **HI-CUT** frequencies.

To **activate** or **deactivate** the EQ for the chosen source, simply press the Equalizer button. To manipulate the **5 frequency bands** (LOW CUT, BASS, BASS-MID, MID, MID-HIGH, HIGH, and HI-CUT), navigate to

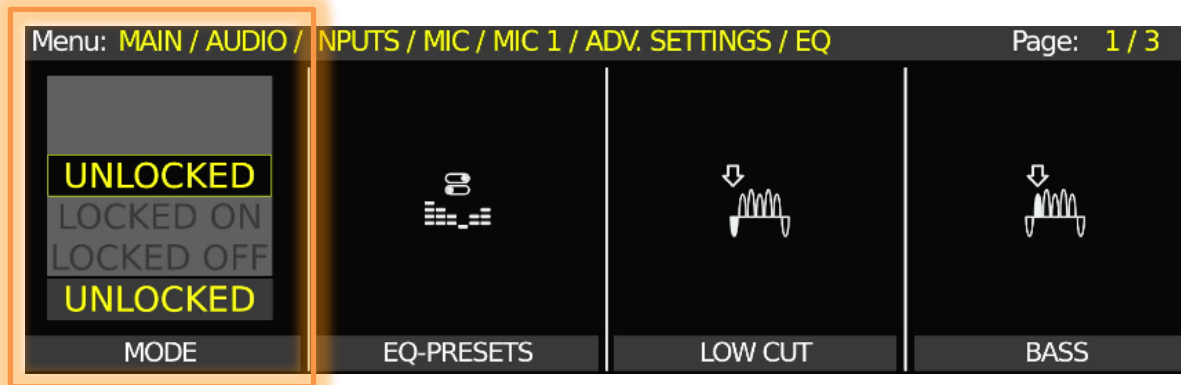
MENU / AUDIO / INPUTS / MIC / MIC 1 / ADV. SETTINGS / EQ

Press the menu button (**«**) or (**»**) to switch between the pages of **EQ** types that are available.

Select the specific frequency to be adjusted with the **FREQUENCY** knob and adjust the bandwidth of the **EQ** with the knob, also you can choose between **PEAK** or **SHELVING** of **BASS** and **HIGH**.

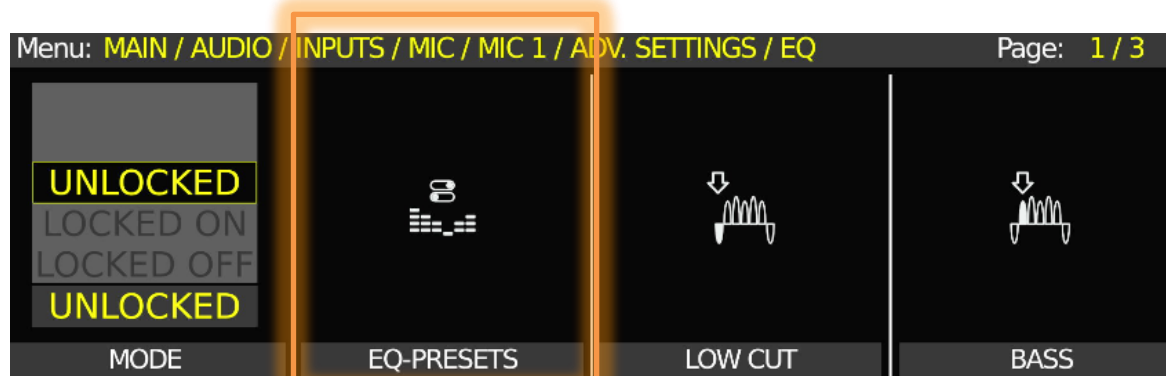
Finally, boost or cut the selected frequency with the **Gain** knob.

After confirming any editing, you will see clearly the change that you do in the graphic equalizer display with different color signals, as well as it is effective at the input source quality and character

MODE:

This function has been added to prevent unauthorized people from activating or deactivating the equalizer.

- **UNLOCKED:** that is mean the equalizer is free and can activate or deactivate the equalizer from the F button on the surface.
- **LOCKED ON:** that is mean the equalizer of this channel will be still **ON** and cannot turn off from the F button.
- **LOCKED OFF:** that is mean the equalizer of this channel will be still **OFF** and cannot turn ON from the F button.

EQ PRESET:

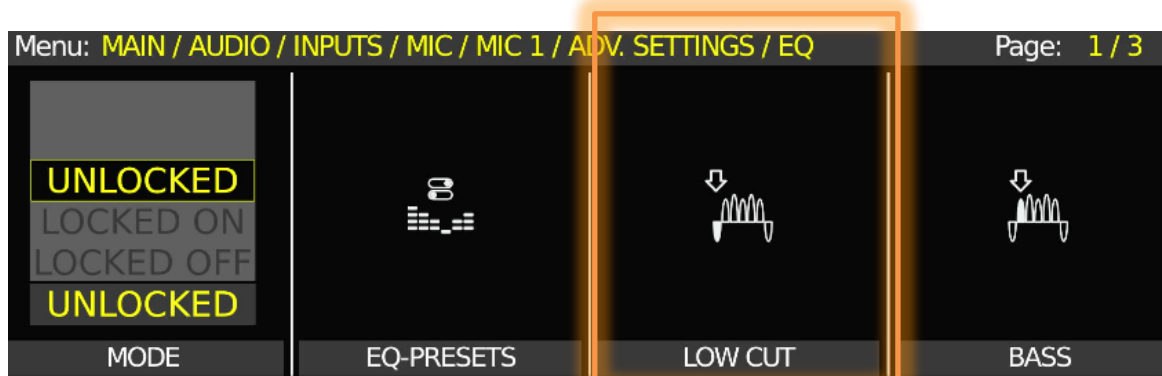
This function is useful to save and recall the equalizer settings. These settings are saved in the internal memory and it is not exportable.

- **SNAPSHOT:** it's able to save ten different **EQ PRESET** by selecting the position of the new save from **1** to **10** before saving it.
- **NAME:** here you can change the name of the PRESET to identify them.
- **RECALL:** to recall one of the PRESETs that are already saved in internal memory.
- **SAVE:** to save the actual equalizer setting as a PRESET.

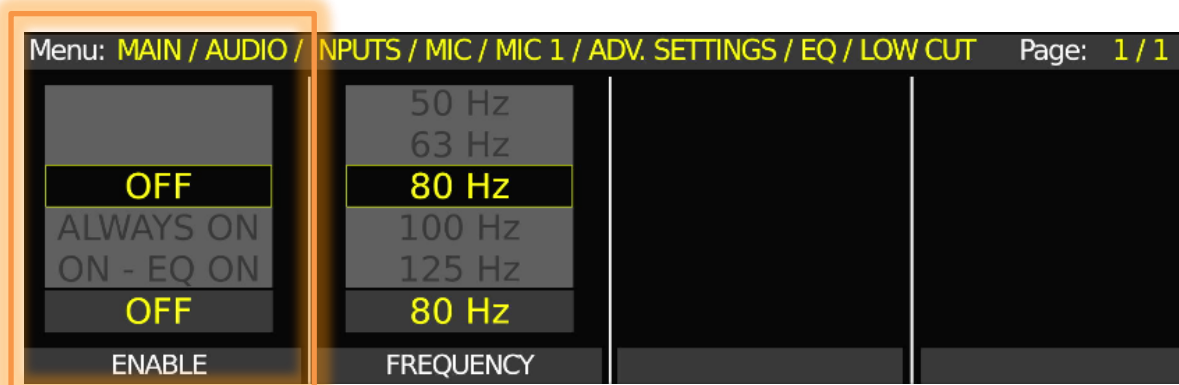
LOW CUT: This is an audio frequency filter that cuts out all of the low sounds below a certain frequency. On the **OXYGEN** console, you are given knobs to control the frequency of the low cut in a full range from **31.5Hz** to **20.0KHz**. Also, you can control how gradual or steep the cut is. Doing so will eliminate the amplification of unwanted low sound. unwanted sound can be:

- Microphone handling or floor rumble (from bumping the mic or stepping on the floor that the mic stand is sitting on).
- The popping of “P’s” and “B’s” from a voice.

- Unwanted electrical hum (60hz).



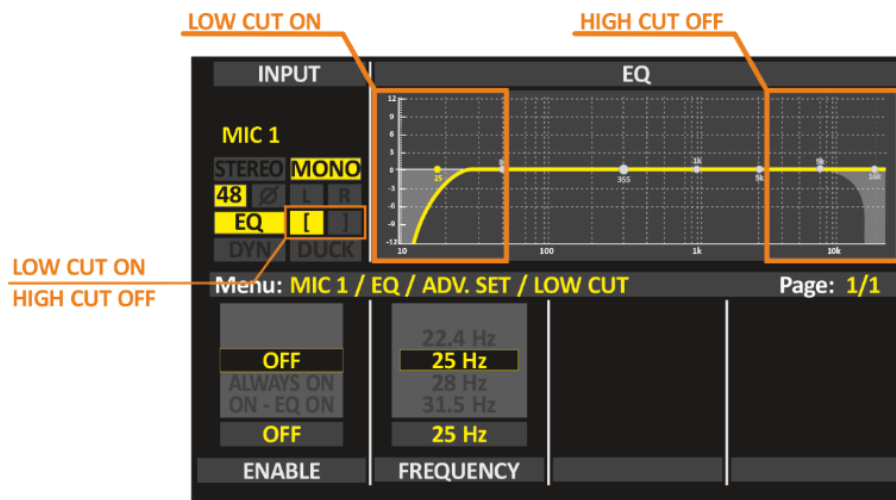
When you activate the LOW CUT function and confirmed it, that change will appear immediately at the output sources, and it will be shown in the EQ graphic display as well the LOWCUT mark ([]) will change to yellow color. Moreover in LOW CUT you could select three different mode: OFF, ALWAYS ON and ON-EQ ON.



- **OFF:** Permanently disables the selected filter. The LOW-CUT or HIGH-CUT filter does not process the signal, regardless of the general EQ status. Select this mode when no frequency filtering is required.
- **ALWAYS ON:** Keeps the selected filter **permanently enabled**, independently from the general EQ status. The filter continues processing the microphone signal even when the other EQ bands are disabled. This mode is useful when LOW-CUT is required continuously to remove rumble or handling noise, or when HIGH-CUT must remain active to reduce hiss and unwanted high-frequency content.
- **ON – EQ ON:** Links the selected filter to the **general EQ status**. The LOW-CUT or HIGH-CUT filter operates only when the channel EQ is enabled. When the EQ is disabled, the filter is bypassed together with the other equalization bands.

HIGH CUT:

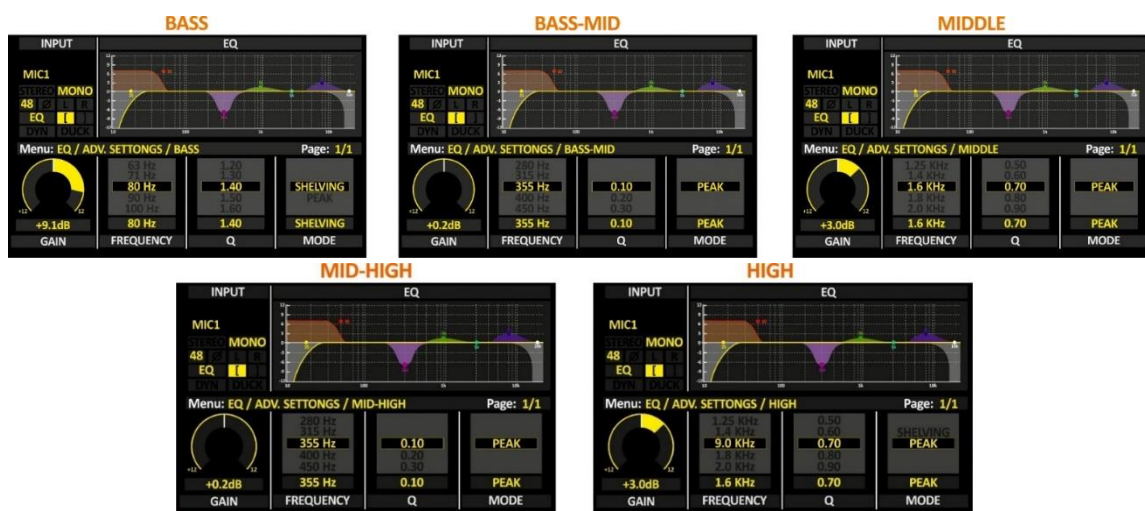
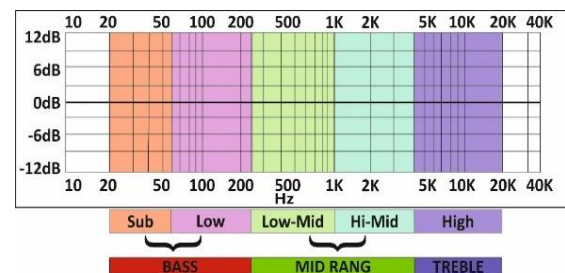
This is an audio frequency filter that cuts out all of the frequency above which the source signals a certain frequency. manipulates the reflection density in the simulated room. As in LOW CUT, in HIGH CUT you could select three different mode: OFF, ALWAYS ON and ON-EQ ON.



BASS, BASS-MID, MIDDLE, MID-HIGH, HIGH:

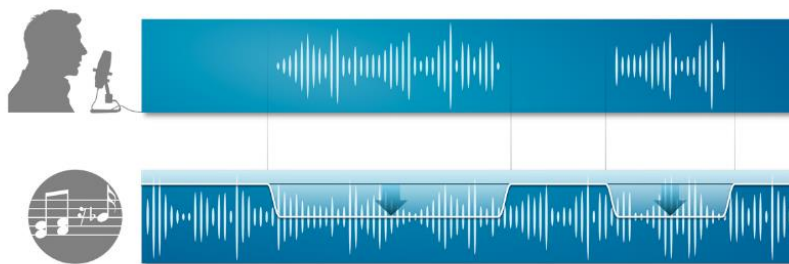
In **OXYGEN** the Parametric equalizers allow the adjustment of multi-band frequency equalizers which allow you to control the three primary parameters: **AMPLITUDE**, **CENTER FREQUENCY**, and **BANDWIDTH**.

The amplitude of each band can be controlled by the **GAIN** knob. The center frequency can be shifted by the **FREQUENCY** knob. Bandwidth (which is inversely related to "Q") can be widened or narrowed by the **Q** knob. **OXYGEN** equalizers can make much more precise adjustments to sound than other equalizers.



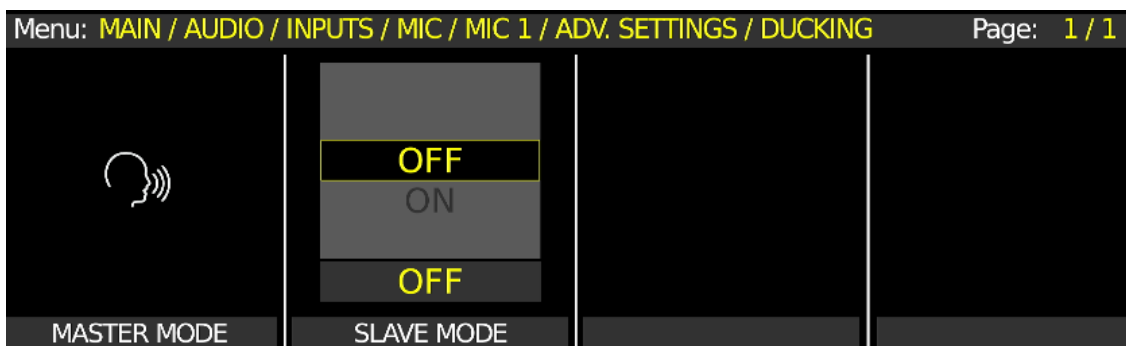
DUCKING

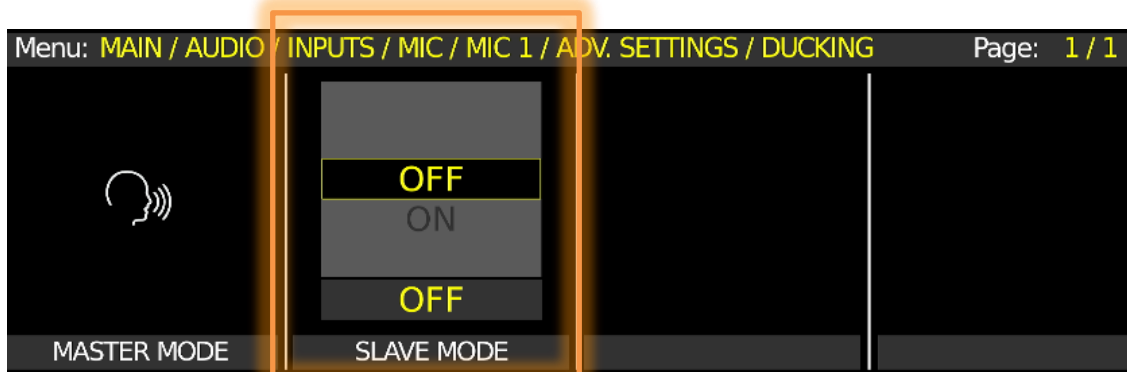
In the musical programs when it is mixed with a **speech** that needs drop music when the anchorman or the guest starts speaking, the background music instantly drops, then it pops right back up again as soon as that person finishes talking. This happens when the **ducking** effect is in action.



Ducking temporarily lowers, or “ducks,” the volume level of a specified audio signal anytime a second specified audio signal is present. In live sound, ducking is commonly used to lower background music anytime a person speaks, then raises it when that person finishes speaking

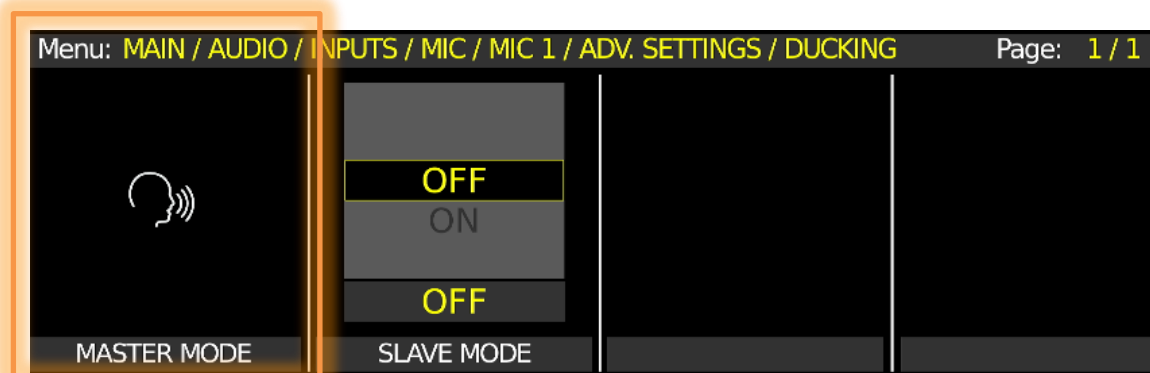
In this page you can set two different mode: **SLAVE MODE** and **MASTER MODE**. If you press the knob on **MASTER MODE** next you will be able to set other functions: **STATUS, THRESHOLD, DUCKING, ATTACK, RELEASE**.



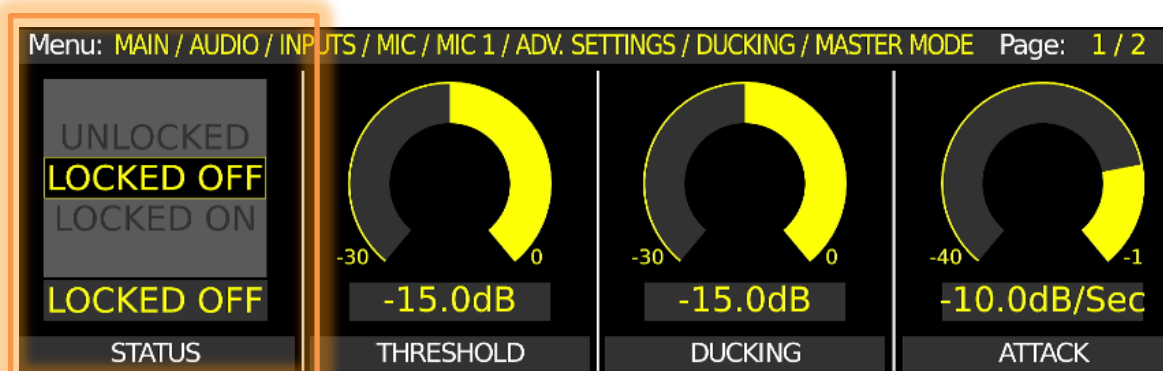
SLAVE MODE:

Determines whether the selected source is affected by the Ducking process. When Slave Mode is enabled, the source is automatically attenuated whenever an active Master source triggers Ducking. When disabled, its level remains unchanged.

- **OFF:** the source is not attenuated by any Ducking Master.
- **ON:** the source is attenuated whenever a Master source activates Ducking.

MASTER MODE:

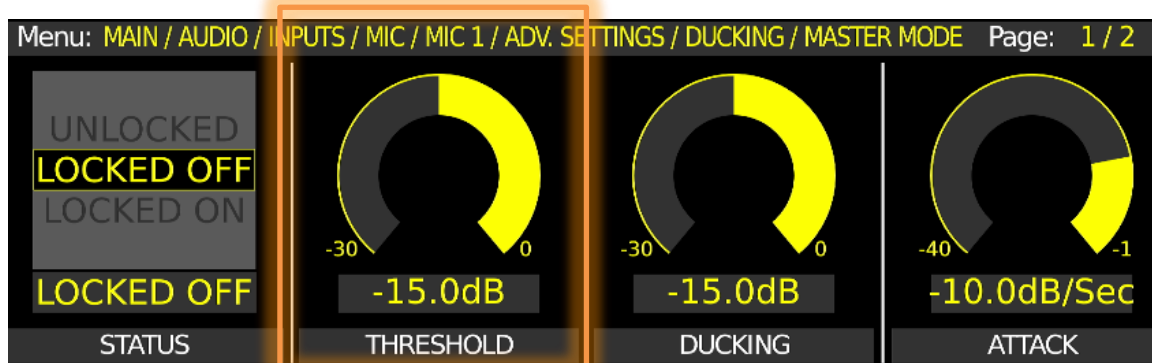
Configures the selected source as a **Ducking control source**. A Master source is not necessarily attenuated; instead, its signal level is monitored and used to control the attenuation of all sources enabled in Slave Mode. For example, MIC 1 can be configured as the Master so that background music is reduced while the presenter is speaking.

STATUS:

Defines whether the selected source can operate as a Ducking Master and whether its operating state can be changed from the console surface.

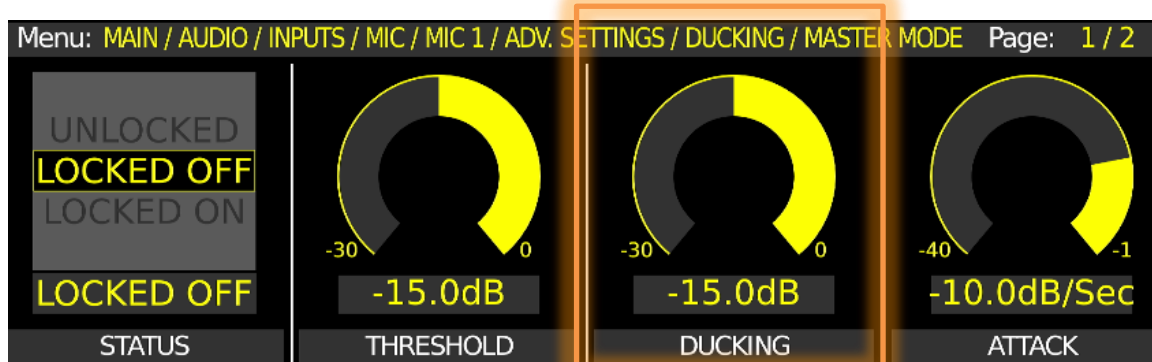
- **LOCKED ON:** the Master function remains **permanently enabled** and cannot be disabled using the assigned F button.
- **LOCKED OFF:** the Master function remains **permanently disabled** and cannot be enabled using the assigned F button.
- **UNLOCKED:** the Master function can be **enabled or disabled from the console surface**, provided that the corresponding F button is configured in **Ducking mode**.

THRESHOLD:



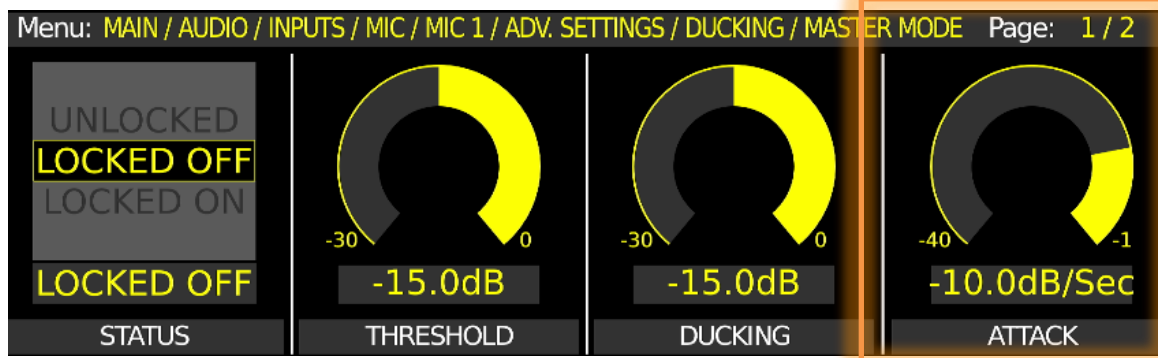
Sets the minimum level that the Master signal must reach to trigger Ducking. When the MIC 1 signal rises above this threshold, the console begins attenuating all sources enabled in Slave Mode. When the signal falls below it, the Ducking release phase begins. A threshold set too low may respond to background noise, while a threshold set too high may not detect quiet speech correctly. **Range: -30.0 dB to 0 dB. Step: 0.1 dB.**

DUCKING:

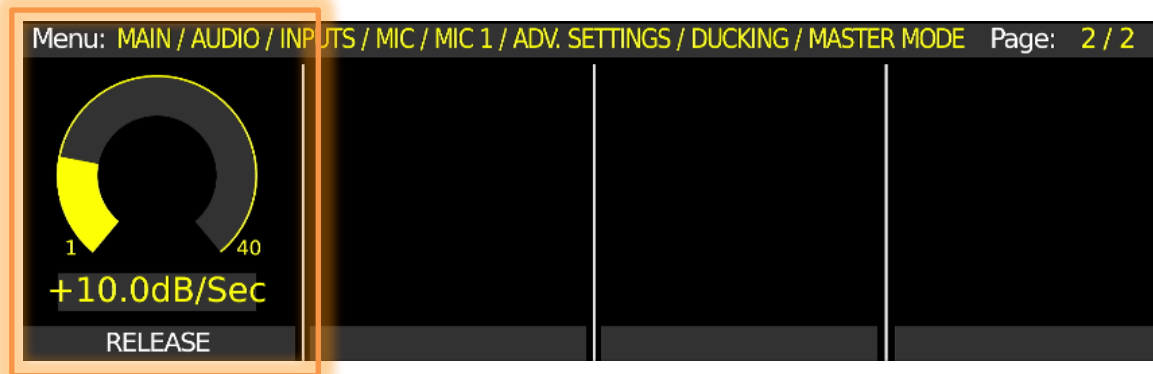


Defines the amount of level reduction applied to all sources enabled in **Slave Mode** when the Master signal activates the process. A greater attenuation value produces a stronger reduction of music or background audio. This parameter affects the Slave sources and **does not reduce the level of the Master source**.

Range: -30.0 dB to 0 dB. Step: 0.1 dB

ATTACK:

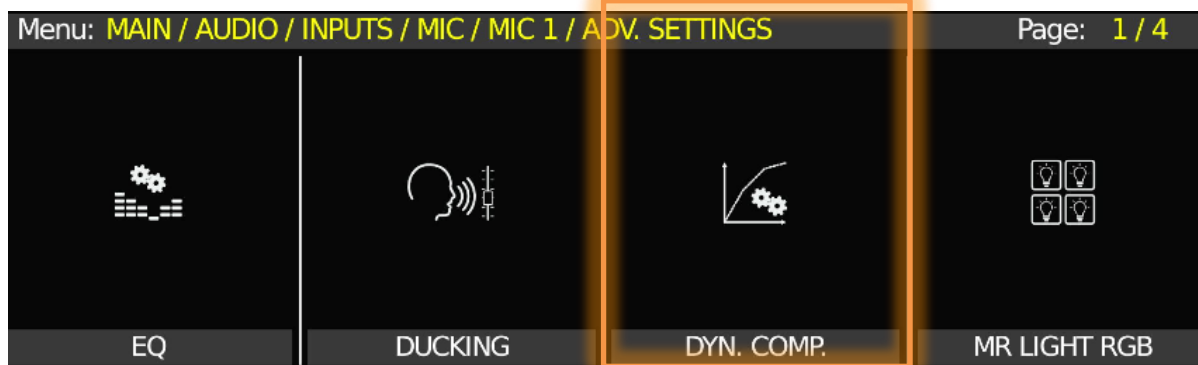
Defines how quickly the Slave sources are reduced from their normal level to the configured Ducking level after the Master signal exceeds the Threshold. A short Attack provides rapid attenuation and improves speech intelligibility, while a longer Attack produces a smoother transition but may allow background audio to mask the beginning of the speech. **Range: -40.0 dB to -1.0 dB. Step: 1.0 dB**

RELEASE:

Defines how quickly the Slave sources return to their original level after the Master signal falls below the Threshold or no longer activates Ducking. A short Release restores the audio rapidly but may produce abrupt level changes. A longer Release provides a **smoother and more natural recovery**, particularly during short pauses in speech. **Range: 1.0 dB/sec to 40.0 dB/sec. Step: 1.0 dB/sec.**

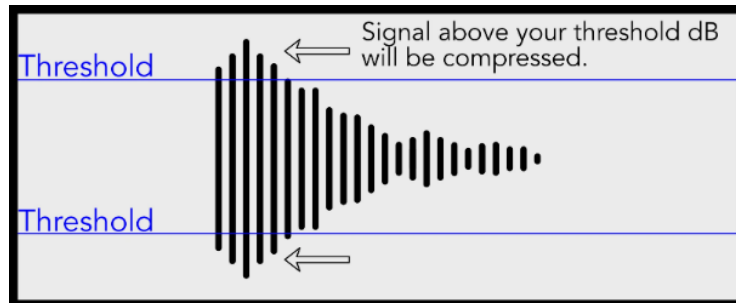
DYN. COMP.

Dynamic range compression (**DRC**) or simply compression is an audio signal processing operation that reduces the volume of loud sounds or amplifies quiet sounds thus reducing or compressing audio signals in **DYNAMIC RANGE**. Compression is commonly used in sound recording and reproduction, broadcasting, live sound reinforcement, and some instrument amplifiers.



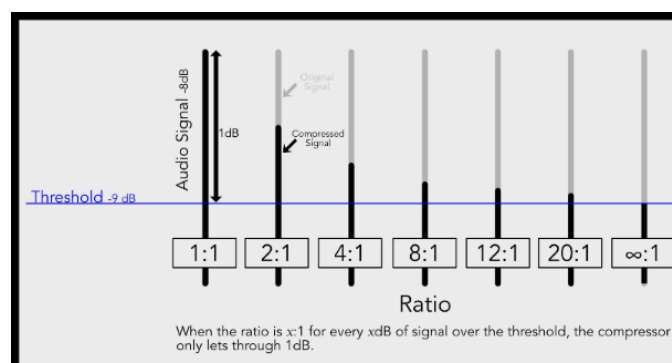
THRESHOLD

A compressor reduces the level of an audio signal if its amplitude exceeds a certain threshold. The threshold is commonly set in decibels (dB), where a lower threshold of **-70 dB** means a larger portion of the signal is treated. When the signal level is below the threshold, no processing is performed and the input signal is passed, unmodified to the output. Thus, a higher threshold of **0 dB**, results in less processing, and less compression.



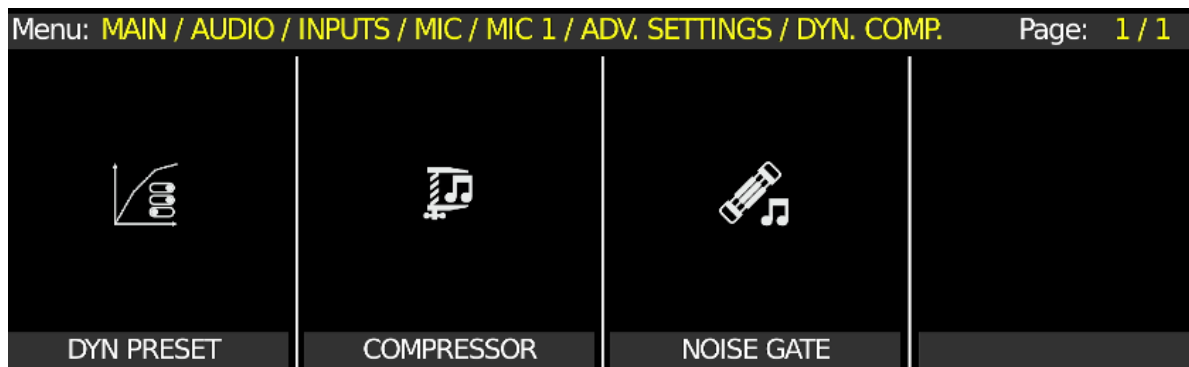
RATIO

When the signal becomes louder than the threshold, the compressor reduces the gain based on the ratio setting. Without getting too mathematical, the ratio of a compressor determines how much gain reduction is applied to a signal after it crosses the threshold.

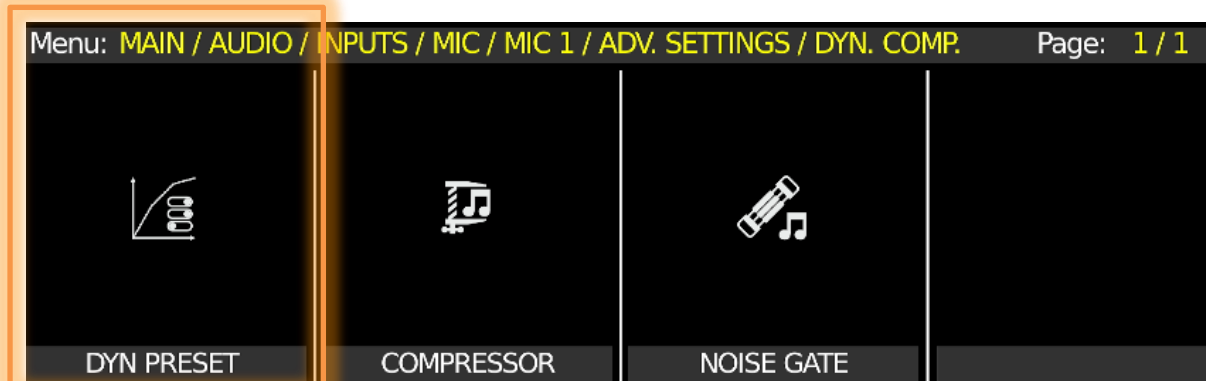


With a ratio of **2:1**, for every **2 dB** above the threshold, the compressor only allows **1 dB** above the threshold through. With a ratio of **4:1**, for every **4 dB** above the threshold, the compressor only allows **1dB** above the threshold through. With a ratio of **10:1**, for every **10 dB** above the threshold, the compressor only allows **1 dB** above the threshold through.

In this page you can select you can select 3 different pic: **DYN PRESET**, **COMPRESSOR** and **NOISE GATE**.

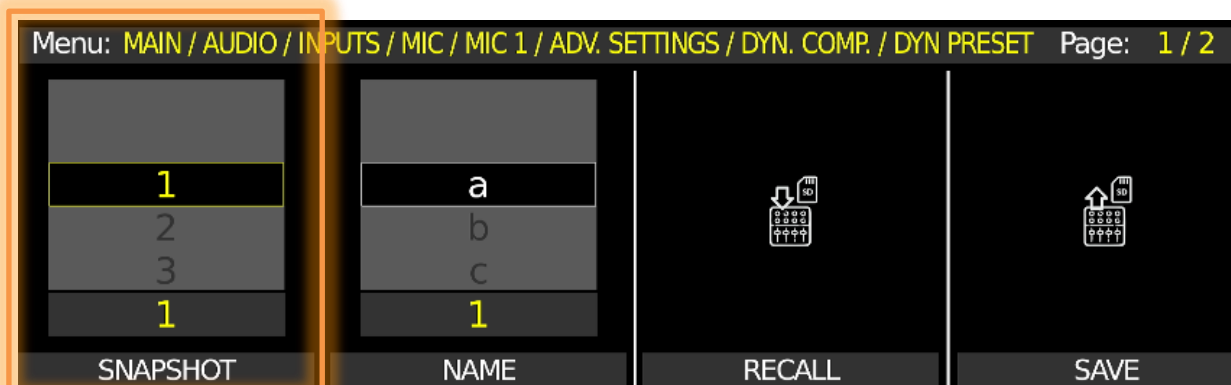


DYN PRESET:



Allows the operator to select and recall a predefined configuration for the channel's **dynamic processing section**. Each preset contains a group of parameters associated with the **Compressor and Noise Gate**, providing a quick and consistent setup for different audio sources, such as speech, microphones or programme material. After selecting a preset, verify that its parameters are suitable for the **level and characteristics of the input signal**.

SNAPSHOTS:

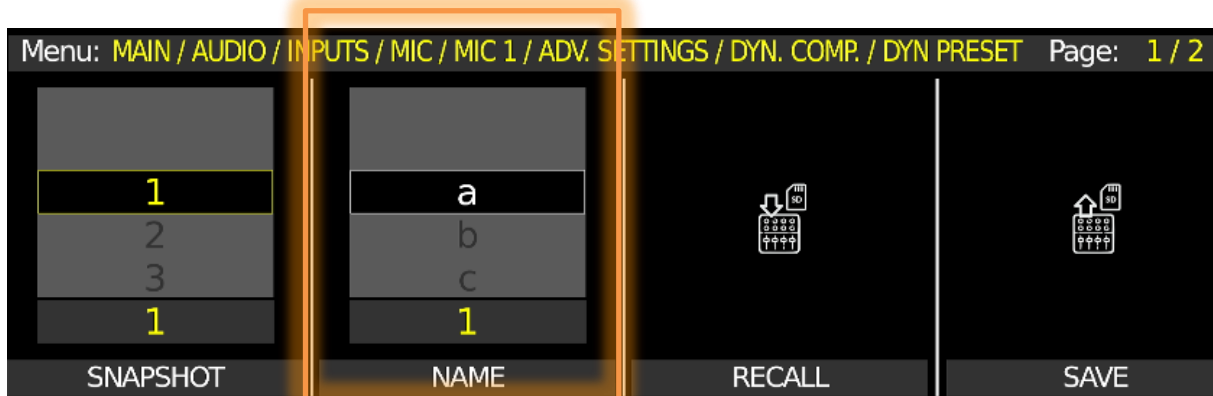


SNAPSHOTS selects the dynamics-preset memory position used for naming, saving or recalling a configuration. **Selectable setting:**

- **SNAPSHOT 1-10**

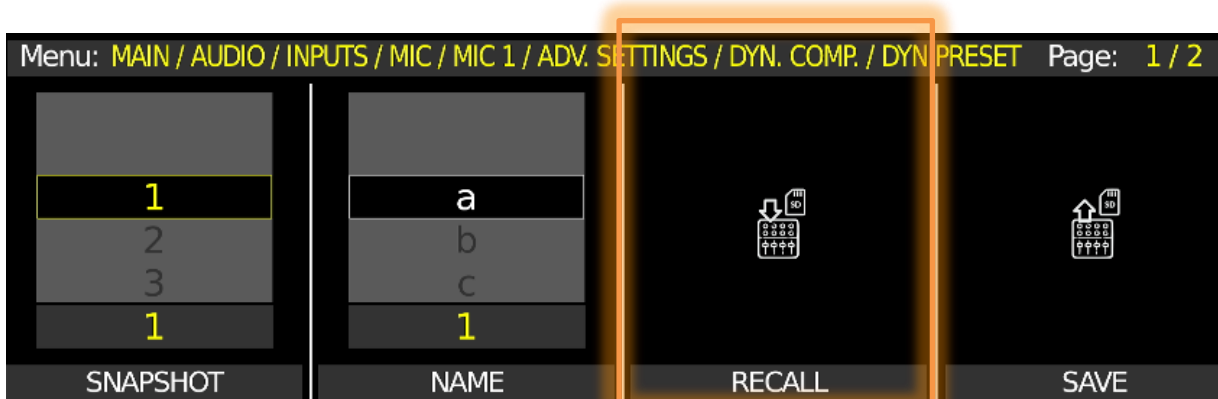
Selecting a position does not necessarily apply it immediately. The operator must normally use **RECALL** to load its settings. Similarly, selecting a position does not save the current settings until **SAVE** is confirmed. Although **SNAPSHOTS** and **SNAPSHOT** refer to the same preset-memory area, they can appear on different pages of the DYN PRESET menu. **Important:** **SNAPSHOTS** selects the memory position. It is not the command that recalls or saves the preset. **Warning:** saving into an occupied position normally overwrites its previous dynamics configuration. Verify the selected position before confirming **SAVE**.

NAME:



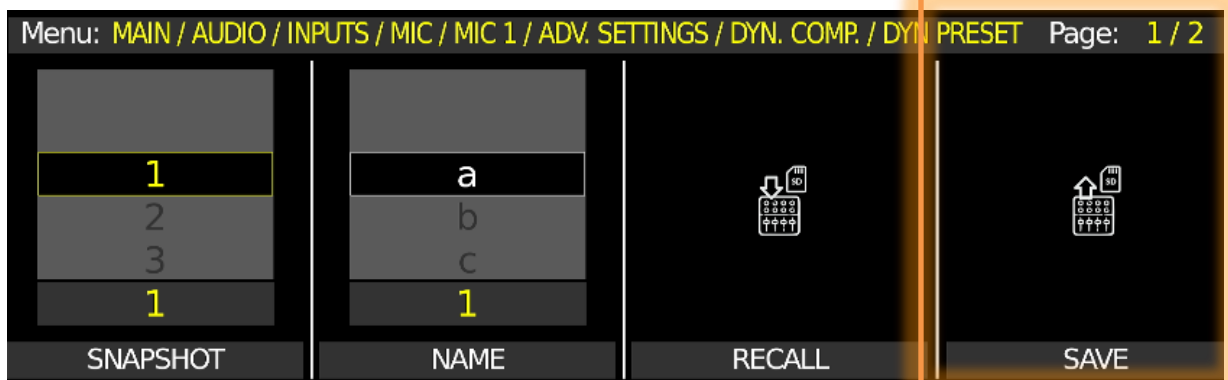
NAME assigns an identifying label to the selected DYN PRESET. The name should clearly identify the intended application of the preset. A clear name helps prevent the wrong dynamics configuration from being recalled during operation. . It is not possible to have a LABEL of more than 8 characters.

RECALL:

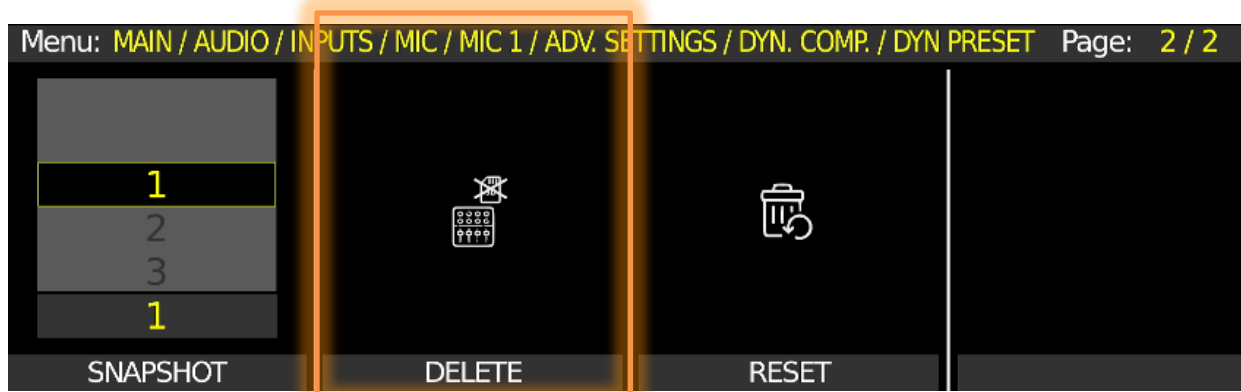


RECALL loads the selected DYN PRESET into the dynamic processor of the current source. When **RECALL** is executed, the stored settings may immediately replace the current values.

Warning: **RECALL** can change the source's audio processing immediately. If the source is On Air, the voice level, compression, noise-gate behaviour and perceived loudness may change without delay.

SAVE:

SAVE stores the current DYN. COMP. parameters in the selected preset position. SAVE captures the current dynamics configuration after the compressor and noise-gate parameters have been adjusted. The saved preset can subsequently be recalled on the same source or another compatible source.

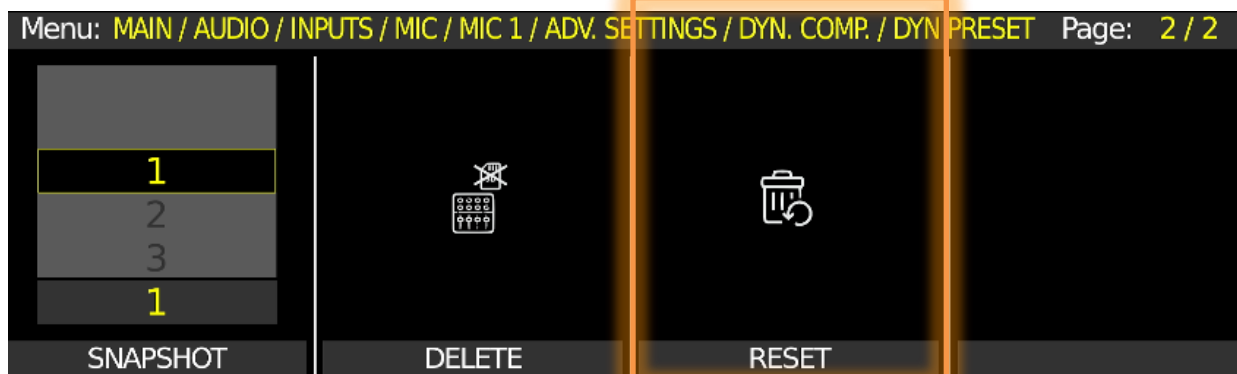
DELETE:

DELETE removes the stored DYN PRESET from the selected SNAPSHOT position. When DELETE is executed:

- The stored preset data is removed
- The preset NAME may be removed or returned to its default label
- The memory position becomes available for a new preset

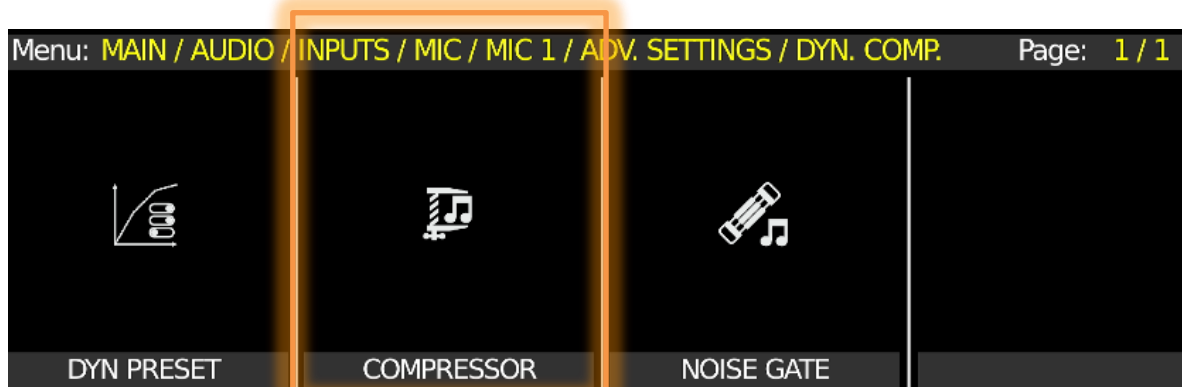
Important: deleting a preset does not normally change the dynamics values already loaded into the active source. The source continues using its current processing until another preset is recalled, the values are manually changed or RESET is executed. **Warning:** DELETE may be irreversible. Confirm that the correct SNAPSHOT position has been selected before proceeding.

RESET:



RESET restores the applicable dynamics configuration to its initial or default state. **Important:** RESET and DELETE are different. DELETE removes a stored preset from memory, while RESET restores the applicable dynamics settings to their default or initial values.

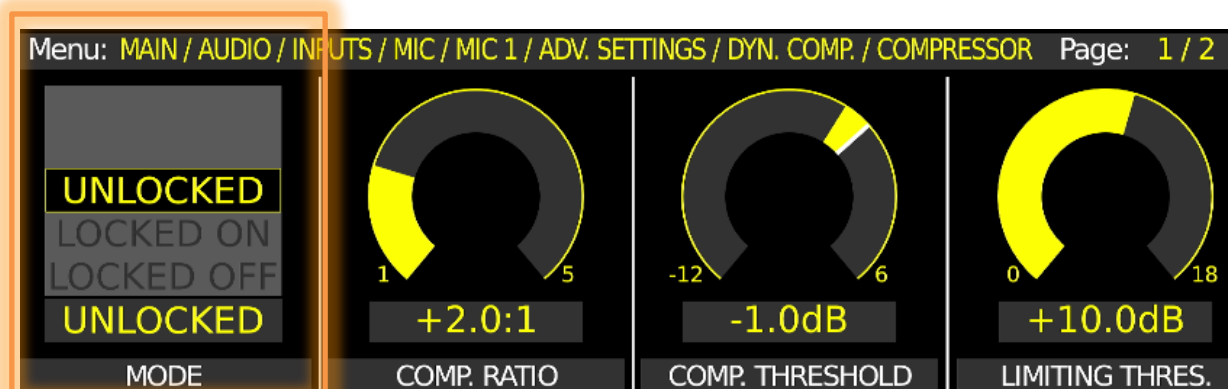
COMPRESSOR:



Controls the **dynamic range** of the input signal by reducing its level when it exceeds the configured **threshold**. The amount of gain reduction is determined by the selected compression parameters, particularly the **ratio, attack and release** settings. Correct configuration helps control sudden peaks and provides a **more stable and consistent output level** without significantly altering the natural character of the audio. In this page you can set different parameters:

MODE:

MODE controls the activation state and surface accessibility of the compressor for the selected source.



The selectable MODE options are:

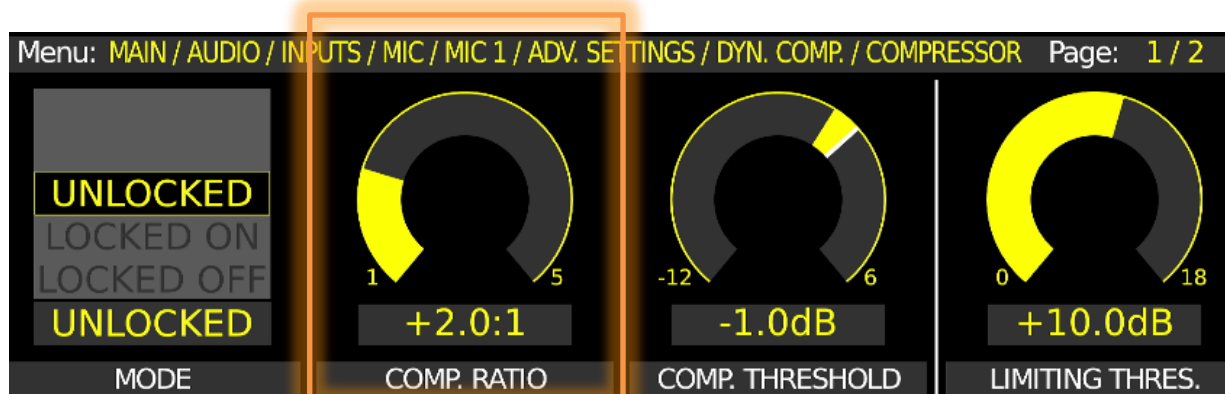
- **UNLOCKED:** When MODE is set to **UNLOCKED**, the compressor is available for normal operator control. If F1 or F2 has been assigned to **COMPRESSOR**, pressing that button can alternate between: Compressor ON and Compressor OFF. For example, a microphone can use: **F1 MODE = COMPRESSOR COMPRESSOR MODE = UNLOCKED**

In this configuration, the operator can press F1 to activate or bypass the compressor.

Important: selecting UNLOCKED does not automatically switch the compressor ON. It makes its ON/OFF state controllable. Always verify the current state before putting the source On Air.

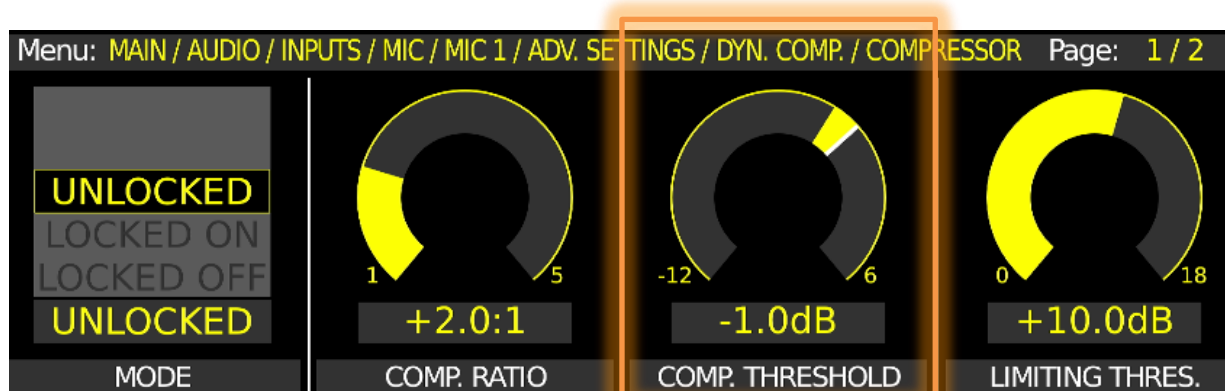
- **LOCKED ON:** When MODE is set to **LOCKED ON**, the compressor remains active and cannot be switched OFF using the F button assigned to COMPRESSOR. LOCKED ON is useful when compression is considered an essential part of the source configuration. This ensures that the configured dynamic control remains active even if the operator presses the associated F button accidentally. **Important:** LOCKED ON does not guarantee that the compressor settings are correct. Excessive compression, an unsuitable threshold or excessive make-up gain can still produce pumping, reduced natural dynamics or clipping at a later stage.
- **LOCKED OFF:** When MODE is set to **LOCKED OFF**, the compressor remains bypassed and cannot be switched ON using the F button assigned to COMPRESSOR. Selecting **LOCKED OFF** prevents the console compressor from being activated accidentally. **Important:** LOCKED OFF does not reset the compressor parameters. THRESHOLD, RATIO, ATTACK, RELEASE and the other stored values remain available and will become effective again if the compressor is subsequently enabled.

COMP. RATIO:



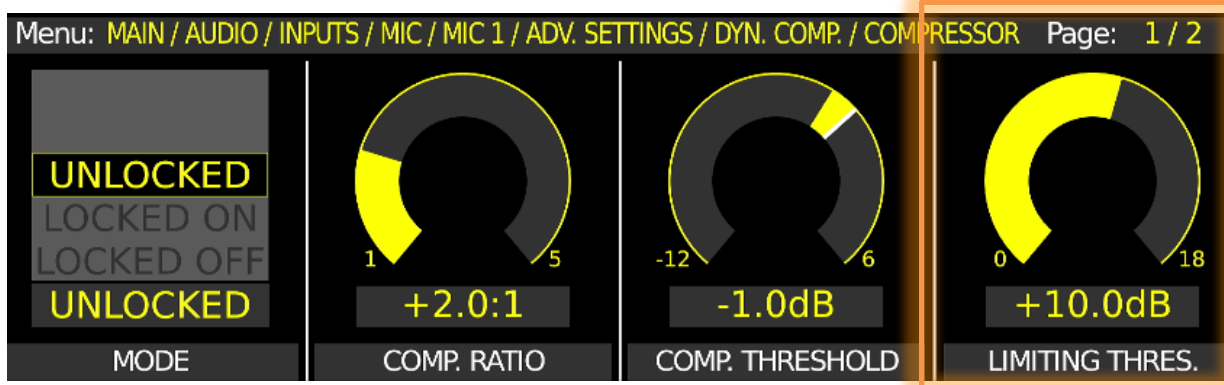
Defines the amount of gain reduction applied when the input signal exceeds the **Compressor Threshold**. For example, with a ratio of **4:1**, every 4 dB above the threshold produces only a 1 dB increase at the output. Higher values provide stronger compression, while **1:1 applies no compression**.

COMP. THRESHOLD:



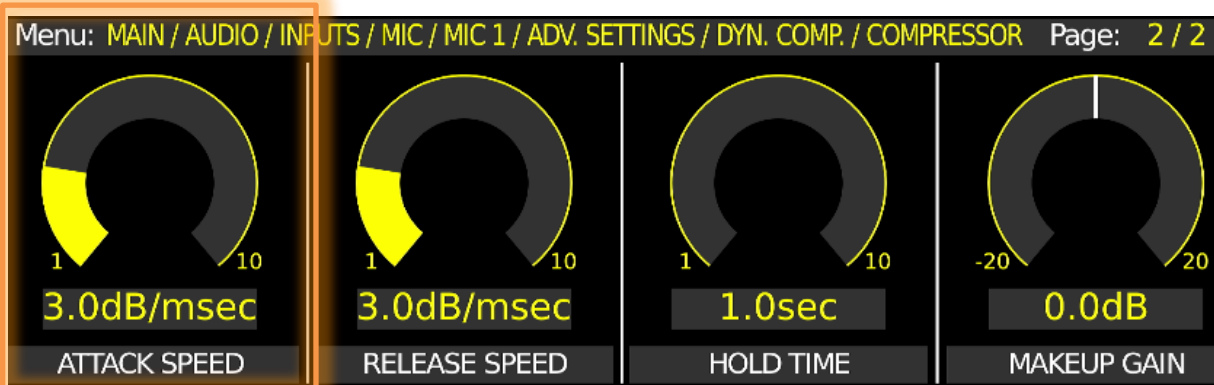
Sets the signal level at which the compressor begins to operate. Signals below the threshold pass without compression, while signals above it are reduced according to the selected **Compression Ratio**. Lower settings apply compression to a larger portion of the input signal. **Range: -12 dB to 6 dB. Step: 0.1 dB.**

LIMITING THRES.:

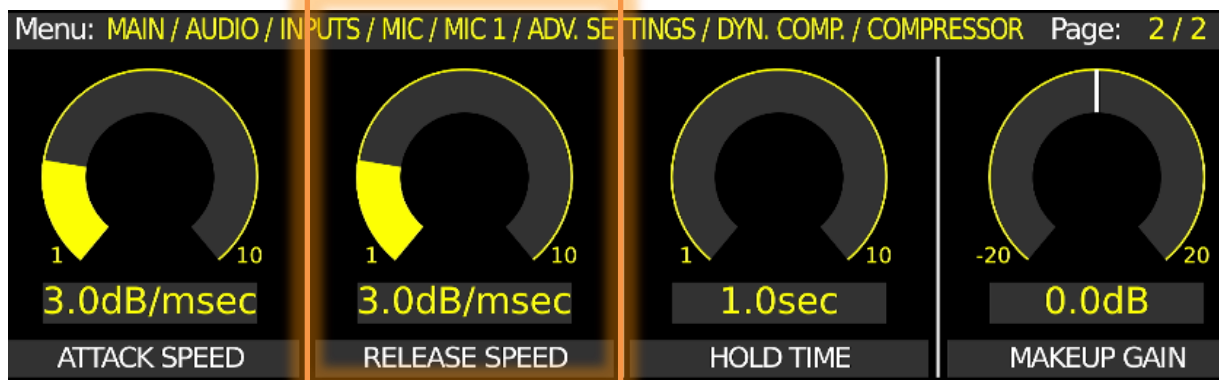


Sets the maximum signal level at which the limiter starts applying strong gain reduction. This function controls high-level peaks and helps prevent **overload, clipping and distortion** in the following stages of the audio chain. A lower setting causes the limiter to intervene earlier. **Range: 0 dB to 18.0 dB. Step: 0.1 dB.**

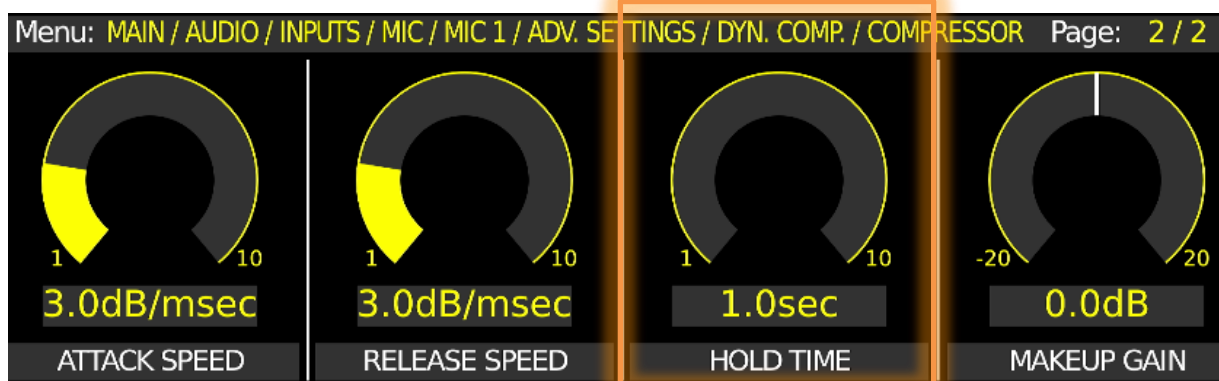
ATTACK SPEED:



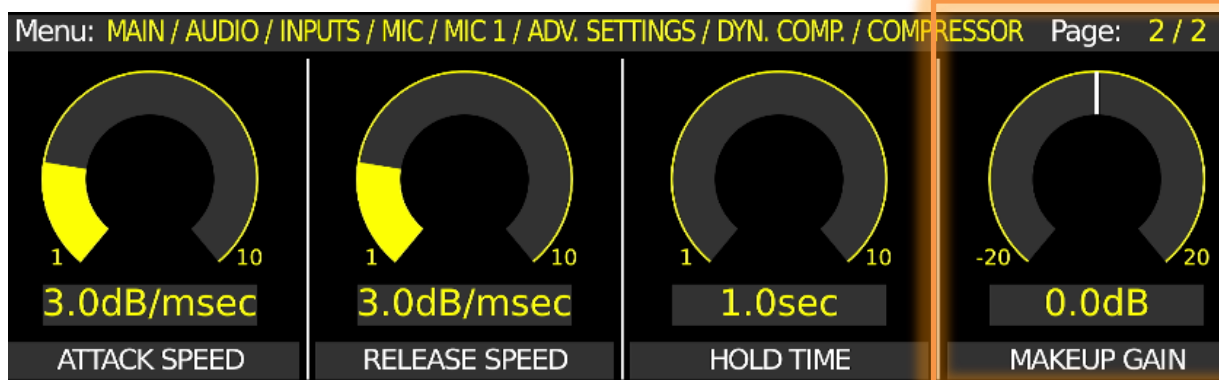
Defines how quickly the compressor reduces the gain after the input signal exceeds the configured threshold. A short attack controls peaks rapidly, while a longer attack preserves more of the signal's initial transient. Excessively short values may make the audio sound less natural. **Range: 1.0 dB/msec to 10.0 dB/msec. Step: 1.0dB**

RELEASE SPEED:

Defines how quickly the compressor returns to its normal gain after the input signal falls below the threshold. A short release provides faster recovery but may produce audible level variations, while a longer release creates smoother and more gradual processing. **Range: 1.0 dB/msec to 10.0 dB/msec. Step: 1.0dB**

HOLD TIME:

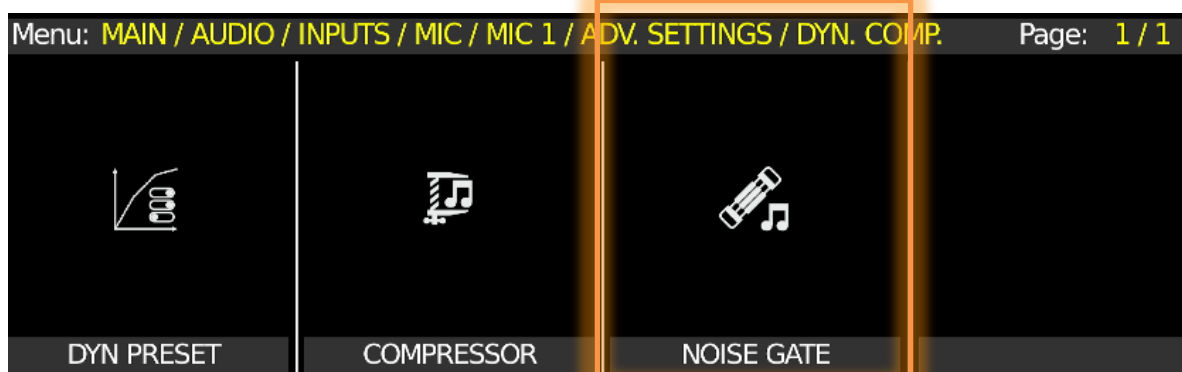
Defines how long the current gain-reduction state is maintained before the processor begins the release phase. This prevents rapid switching when the signal fluctuates around the threshold and helps provide **more stable dynamic processing**. **Range: 1.0 sec to 10.0 sec. Step: 1.0 sec.**

MAKEUP GAIN:

Applies gain after compression to compensate for the level reduction introduced by the compressor. It allows the processed signal to reach an appropriate output level while preserving the controlled dynamic range.

Excessive Makeup Gain may cause the signal to reach the limiter or produce **clipping**. **Range: -20.0 dB to +20.0 dB. Step: 0.1dB**

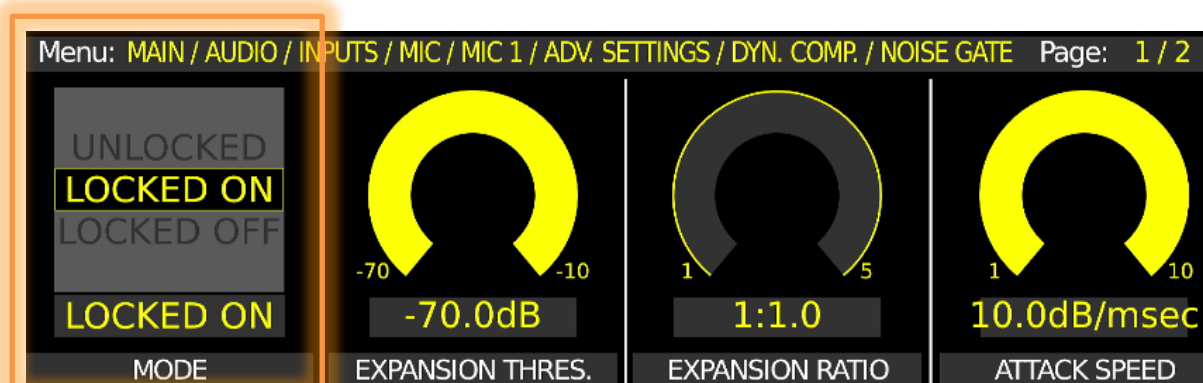
NOISE GATE:



Attenuates or closes the channel signal when its level falls below the configured **threshold**. This helps reduce **background noise, microphone hiss and unwanted ambient sound** during silent periods. When the input signal rises above the threshold, the gate opens and allows the audio to pass through. If the threshold is set too high, **quiet speech or the beginning of a sentence may be unintentionally attenuated**. In this page you can set different parameters:

MODE:

MODE controls the activation state and surface accessibility of the **NOISE GATE** for the selected source.



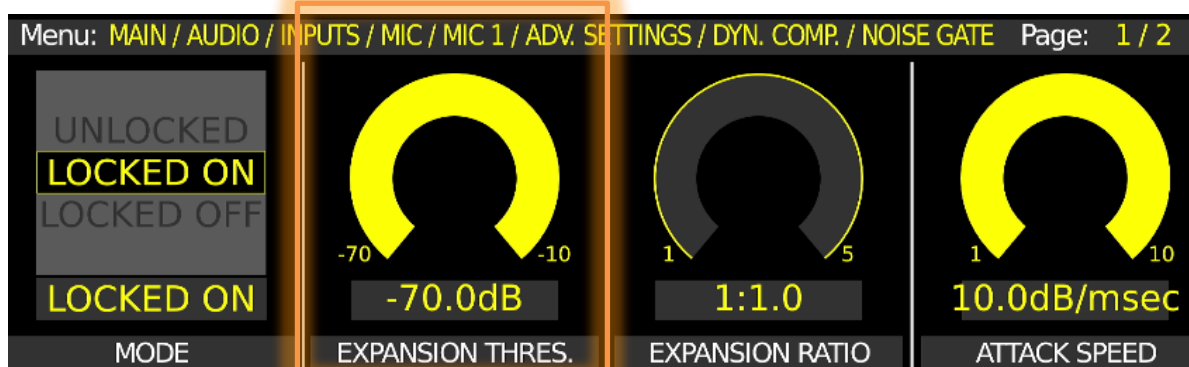
The selectable **MODE** options are:

- **UNLOCKED:** When MODE is set to **UNLOCKED**, the NOISE GATE is available for normal operator control. If F1 or F2 is assigned to **NOISE GATE**, pressing the button can alternate between: Noise gate ON and Noise gate OFF. For example, a microphone can use: **F1 MODE = NOISE GATE -- NOISE GATE MODE = UNLOCKED**. In this configuration, pressing F1 activates or bypasses the gate. **Important:** selecting UNLOCKED does not automatically activate the NOISE GATE. It only allows its state to be changed. Always verify whether the processor is currently ON or OFF.
- **LOCKED ON:** When MODE is set to **LOCKED ON**, the NOISE GATE remains active and cannot be switched OFF using the assigned F button. **Important:** LOCKED ON does not mean that the source is permanently muted. The gate attenuates the signal only when the audio falls below the configured EXPANSION THRES. When the signal exceeds the threshold, it is allowed to pass according to the configured attack and expansion behaviour.
- **LOCKED OFF:** When MODE is set to **LOCKED OFF**, the NOISE GATE remains bypassed and cannot be switched ON using the assigned F button. For example, an ambient microphone intended to reproduce the continuous sound of a studio, venue or audience may use: **NOISE GATE MODE =**

LOCKED OFF. This prevents quiet ambience from being removed during pauses. **Important:** LOCKED OFF does not reset EXPANSION THRES., EXPANSION RATIO, ATTACK, RELEASE or HOLD. These parameters remain stored and become effective again if the gate is subsequently enabled.

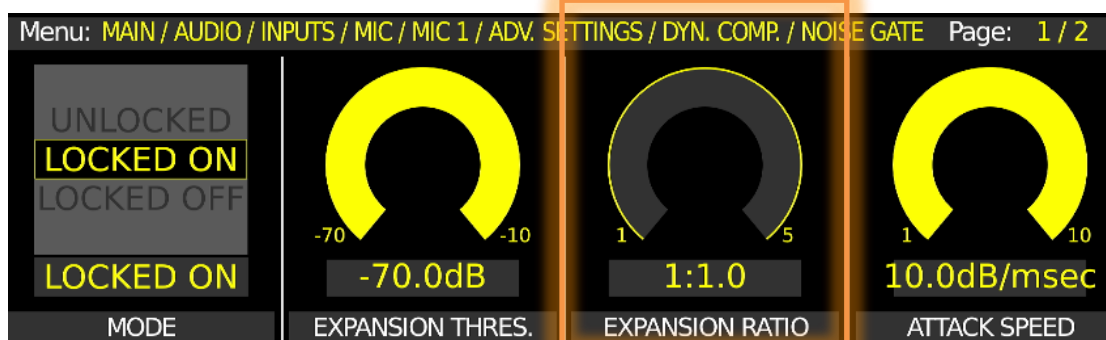
Important: MODE is associated with the selected source rather than only with the physical fader. When switching between CH-A and CH-B, the noise-gate state and F-button behaviour may change according to the settings stored for the newly selected source.

EXPANSION THRES.:



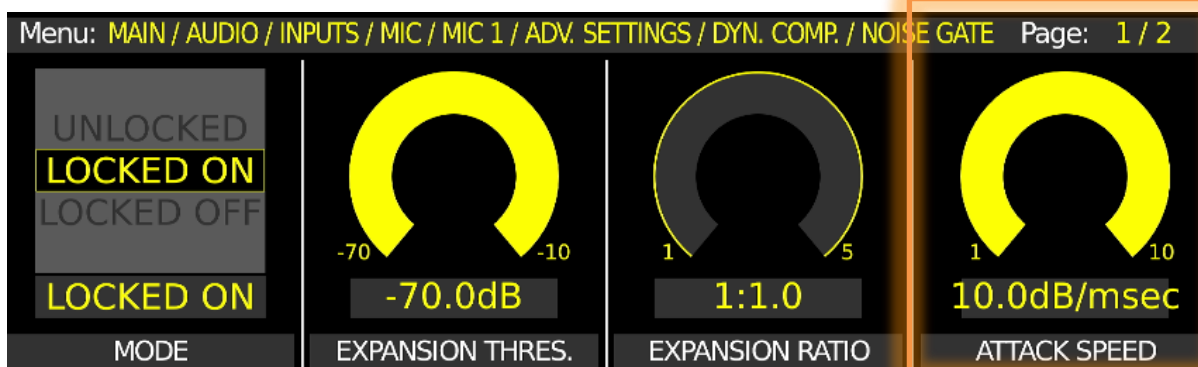
Sets the signal level below which the Noise Gate begins to attenuate the input signal. Signals above the threshold pass normally, while signals below it are progressively reduced according to the configured **Expansion Ratio**. A higher threshold provides stronger noise suppression but may also attenuate low-level speech or other wanted signals. **Range: -70 dB to -10 dB. Step: 0.1 dB.**

EXPANSION RATIO:



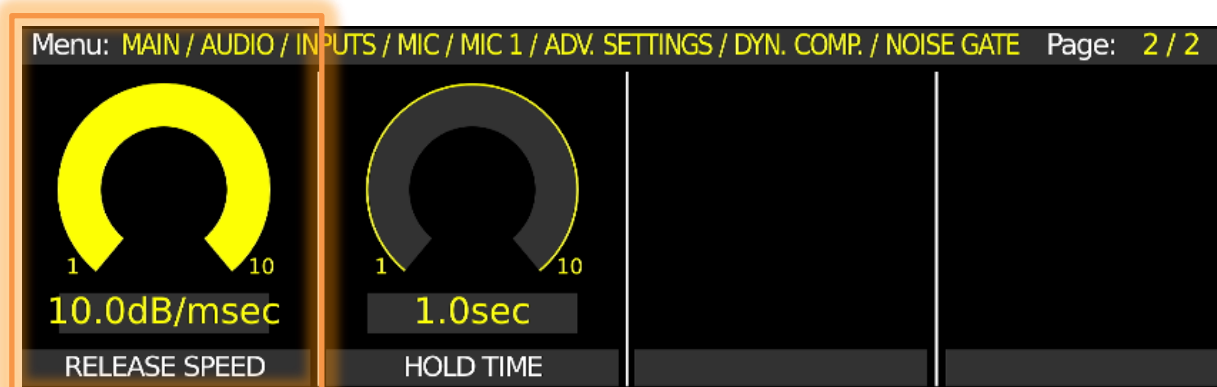
Defines the amount of attenuation applied when the input signal falls below the **Expansion Threshold**. A low ratio produces gradual and natural attenuation, while a higher ratio provides a stronger gating effect. Excessively high values may cause abrupt signal interruptions or make quiet speech sound unnatural.

ATTACK SPEED:



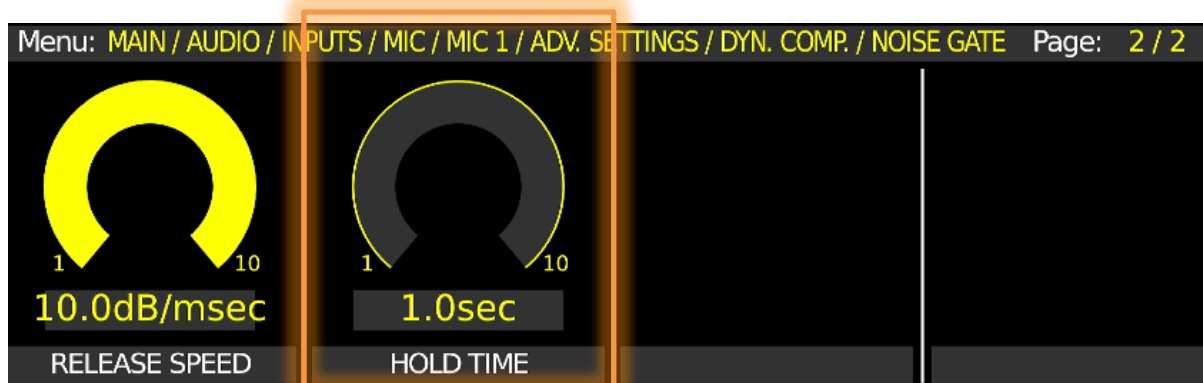
Defines how quickly the Noise Gate opens when the input signal rises above the Expansion Threshold. A short attack allows the signal to pass almost immediately, preserving the beginning of speech and fast transients. A longer attack produces a smoother opening but may attenuate the initial part of the wanted signal. **Range: 1.0 dB/msec to 10.0 dB/msec. Step: 1.0Db**

RELEASE SPEED:



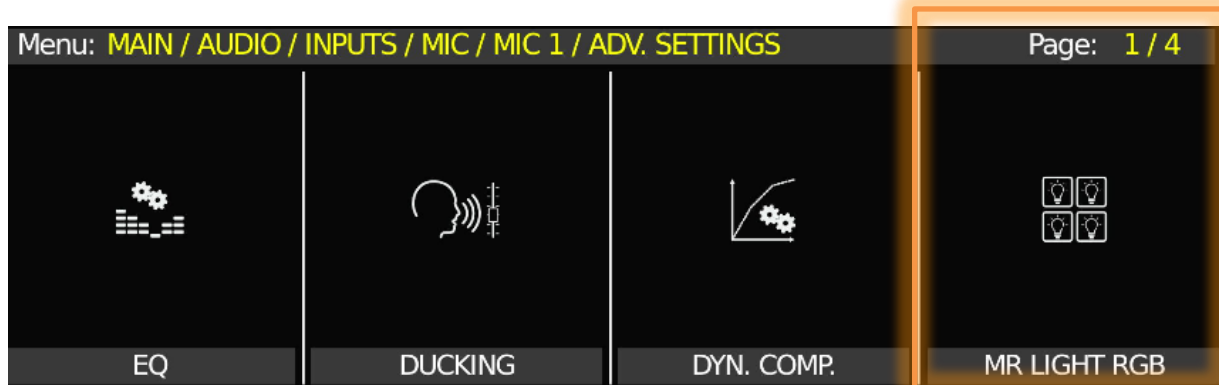
Defines how quickly the Noise Gate attenuates the signal after its level falls below the Expansion Threshold. A short release closes the gate rapidly but may produce abrupt level changes. A longer release provides a smoother and more natural fade-out while allowing more background noise to remain audible temporarily. **Range: 1.0 dB/msec to 10.0 dB/msec. Step: 1.0 dB.**

HOLD TIME:



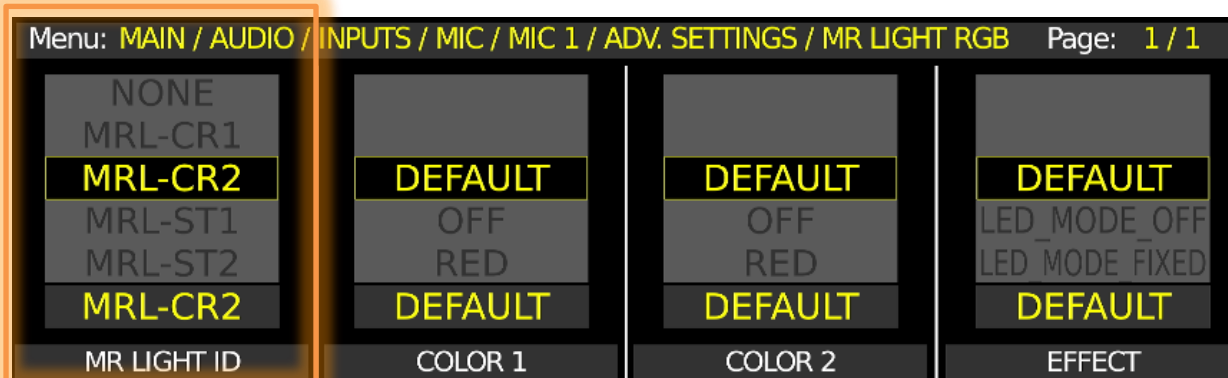
Defines how long the Noise Gate remains open after the input signal falls below the Expansion Threshold and before the release phase begins. This prevents the gate from opening and closing repeatedly during short pauses or small signal-level variations. A suitable Hold Time provides **more stable and natural processing**, particularly for speech. **Range: 1.0 sec to 10.0 sec. Step: 1.0 sec.**

MR LIGHT RGB:



This function enables integration with the **AxelTech MR. LIGHT RGB signalling lamp**. The console can control the lamp's **activation, colour, and operating mode** according to specific events—for example, switching on an **“ON AIR” indication** when a microphone is opened. Depending on the system version, communication may be managed through the **local network or Wi-Fi**.

MR LIGHT ID:



MR LIGHT ID selects the MR Light RGB device that receives the command generated by MIC 1.

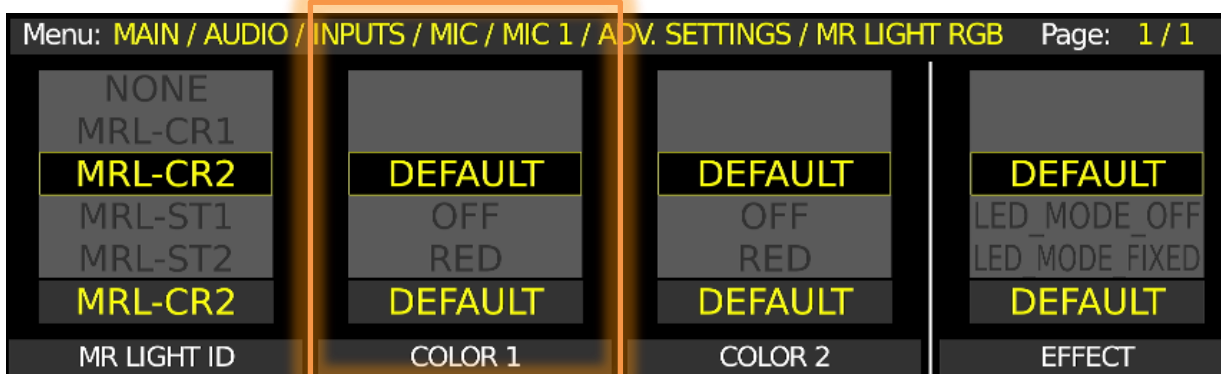
The selectable identifiers are:

- **NONE:** When **NONE** is selected, MIC 1 does not directly control an MR Light RGB device. NONE may be selected when: MIC 1 does not require individual visual signalling / The lamp is controlled through CR LIGHT or ST LIGHT logic / The MR Light RGB installation has not yet been completed
- **MRL-CR1 / MRL-CR2:** **MRL-CR1** and **MRL-CR2** identify the two device slots conventionally assigned to the Control Room. The slot name does not by itself determine the operating logic. The selected slot must also be correctly configured on the general MR LIGHT RGB page. When the device's CONTROL parameter is set to **ON**, MIC 1 can control the corresponding lamp using its individual colour and effect settings. When CONTROL is set to **CR LIGHT**, the device instead follows the general Control Room light command.

- **MRL-ST1 / MRL-ST2:** **MRL-ST1** and **MRL-ST2** identify the two device slots conventionally assigned to the Studio. When the selected device is configured with **CONTROL = ON**, it can respond directly to the individual MIC 1 configuration. When it is configured as **ST LIGHT**, it follows the general Studio light command instead.
- **MRL-1-8:** **MRL-1** through **MRL-8** are general-purpose device slots. They can be used for: Individual microphone indicators / Presenter or guest signalling / Additional On Air lamps / Production-area indicators

These identifiers correspond to the device slots available on the MR LIGHT RGB general configuration page. Each slot must be associated with the network parameters of the required lamp, including its IP address and, where provided, its MAC address and device name.

COLOR 1:

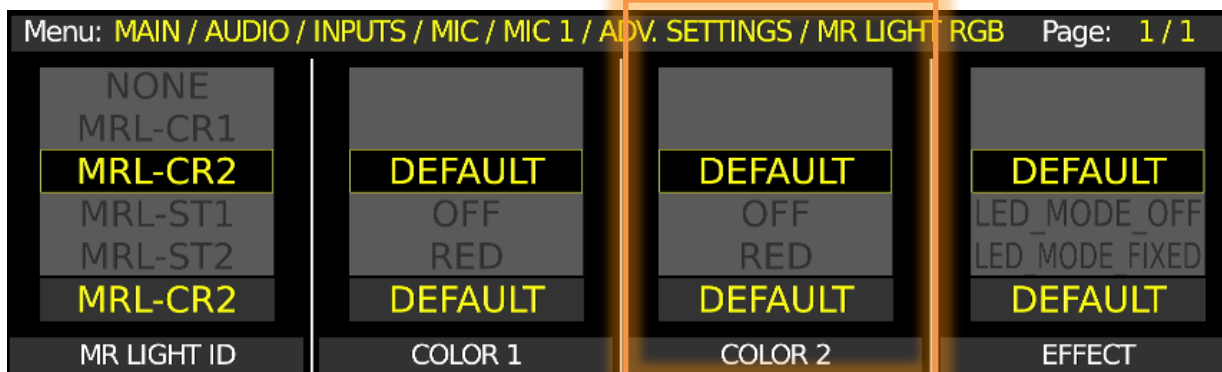


COLOR 1 selects the primary colour used by the MR Light RGB effect assigned to MIC 1.

The available selections are:

- **OFF:** **OFF** represents an unlit colour state. When **COLOR 1** is set to **OFF**: The primary phase of the effect produces no illumination / A **FIXED** effect may leave the lamp unlit
- **RED**
- **GREEN**
- **BLUE**
- **YELLOW**
- **CYAN**
- **MAGENTA**
- **WHITE**

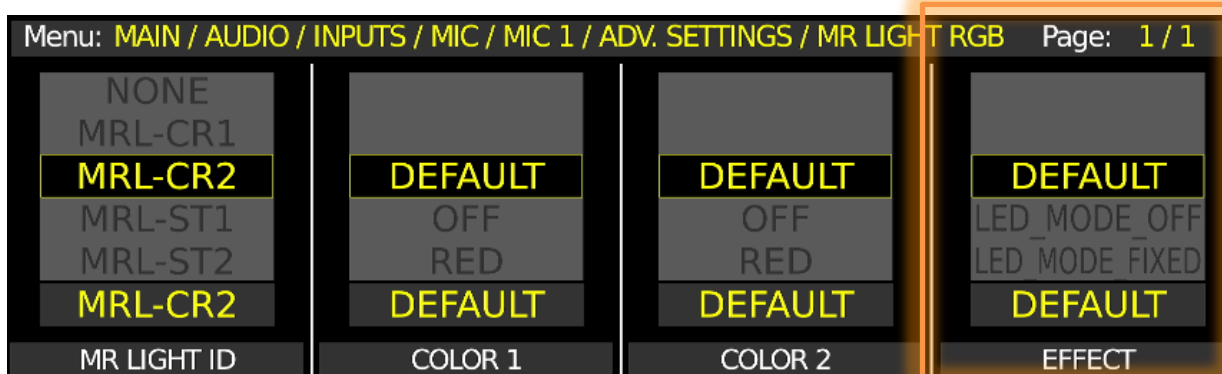
The selected colour is produced by the RGB LEDs installed inside the lamp. Its perceived appearance can be affected by ambient lighting, the printed lamp panel and the configured LED brightness.

COLOR 2:

COLOR 2 selects the secondary colour used by lighting effects that require two colour states.

The selectable values are:

- **OFF**
- **RED**
- **GREEN**
- **BLUE**
- **YELLOW**
- **CYAN**
- **MAGENTA**
- **WHITE**

EFFECT:

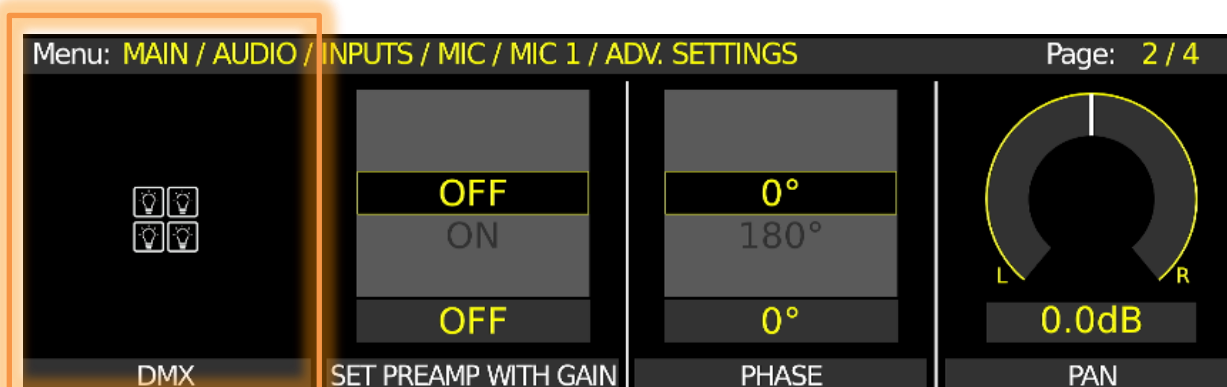
EFFECT determines how COLOR 1 and, where applicable, COLOR 2 are reproduced by the selected MR Light RGB device. The selectable EFFECT options are:

- **LED MODE OFF:** disables the LED output for the selected effect.
In this mode: The lamp remains unlit / COLOR 1 and COLOR 2 do not produce a visible effect / No animation or transition is applied.
This option is useful when the MR LIGHT assignment must remain stored but the lamp must not illuminate.
- **LED MODE FIXED:** produces a continuous, steady indication. In this mode: COLOR 1 remains continuously illuminated / No flashing or movement is applied

- **LED_MODE_BLINK:** produces a periodic flashing indication. Depending on the colour configuration: COLOR 1 may alternate with an unlit phase / COLOR 1 and COLOR 2 may alternate when two-colour operation is supported
- **LED_MODE_SINGLE_WAVE:** produces a single moving or sweeping light pattern across the illuminated area. The primary visible colour is determined by COLOR 1. COLOR 2 may be used as a secondary or background state according to the lamp firmware.
- **LED_MODE_DOUBLE_WAVE:** produces two simultaneous or sequential wave movements.
- **LED_MODE_FADING_WAVE:** **LED_MODE_FADING_WAVE** combines a moving wave with gradual brightness transitions. The illuminated area may progressively increase and decrease in intensity as it moves across the lamp. COLOR 1 is normally the principal colour, while COLOR 2 may define a secondary or background state.
- **LED_MODE_FADING_BLINK:** produces a flashing effect with gradual fade-in and fade-out transitions instead of abrupt switching.
- **LED_MODE_CIRCLE_WAVE:** produces a circular or ring-shaped wave movement. The light may appear to travel around the illuminated area in a circular sequence. COLOR 1 normally defines the principal wave colour, while COLOR 2 may be used as a secondary phase or background.
- **LED_MODE_NOISE:** produces an irregular or pseudo-random animated lighting pattern. This effect is visually more dynamic than fixed or regular flashing and should normally be reserved for special signalling or demonstration purposes.
- **LED_MODE_RAINBOW:** produces a continuously changing multicolour RGB sequence. The identifier uses the firmware spelling **RAIMBOW**.
- **LED_MODE_RAINBOW_WAVE:** combines a multicolour spectrum with a wave or sweeping movement. The lamp may display changing colours while the illuminated pattern travels across the device.
- **LED_MODE_RAINBOW_2:** is a second rainbow-style multicolour mode. It may use a different colour progression, timing, direction or spatial distribution from **LED_MODE_RAINBOW**.
- **LED_MODE_KITT:** produces a scanning effect similar to a moving light that travels across the lamp and returns in the opposite direction.
- **LED_MODE_CYRCLE:** produces a circular or cyclical lighting pattern.

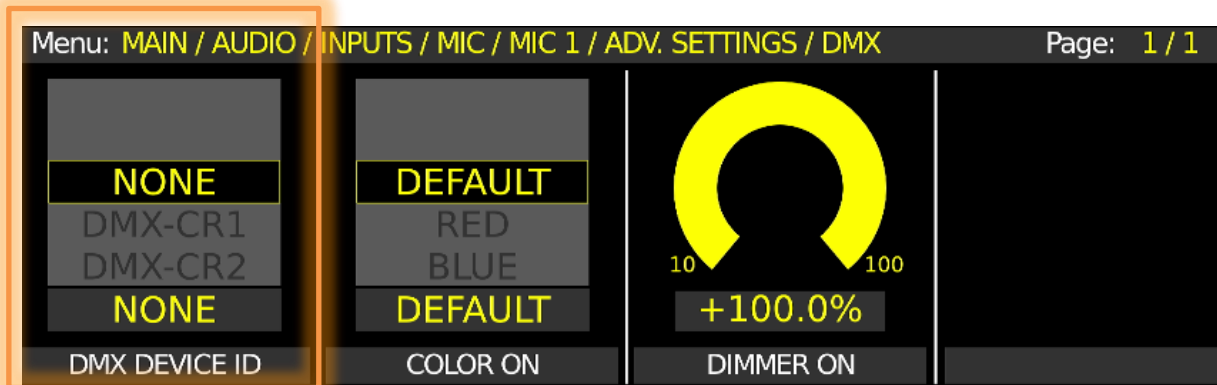
The following descriptions explain the intended operating behaviour suggested by the firmware identifiers. The exact speed, direction, timing and colour handling are determined by the installed MR Light RGB firmware.

DMX



This function enables communication with **DMX-compatible lighting equipment**. It allows the Oxygen 3000 to control studio lights automatically according to console events, such as **microphone activation**, **recording status**, or **On-Air conditions**. Compatible DMX hardware and proper system configuration are required.

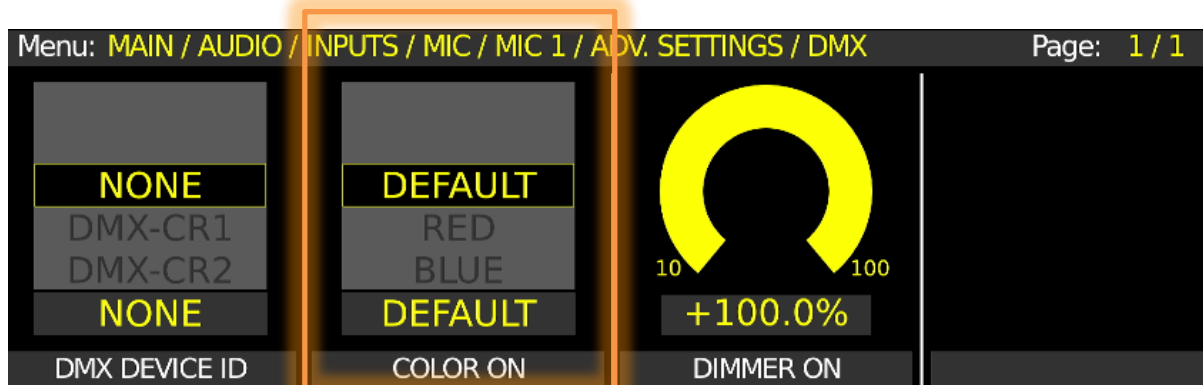
DMX DEVICE ID:



DMX DEVICE ID selects the external DMX lighting device that must respond to the ON state of MIC 1. The selectable values include one of the DMX devices automatically detected by the console. The exact list depends on the devices currently configured and available to the console.

Important: DMX DEVICE ID is the identifier used by the OXYGEN 3000 Plus to select a configured device. It should not automatically be interpreted as the fixture's DMX starting-channel address unless the installed firmware or DMX controller explicitly uses the same value for both functions.

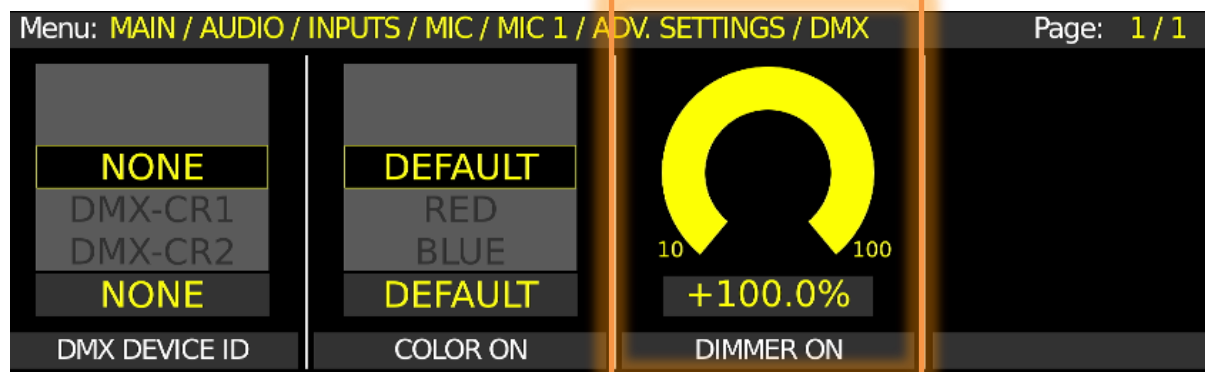
COLOR ON:



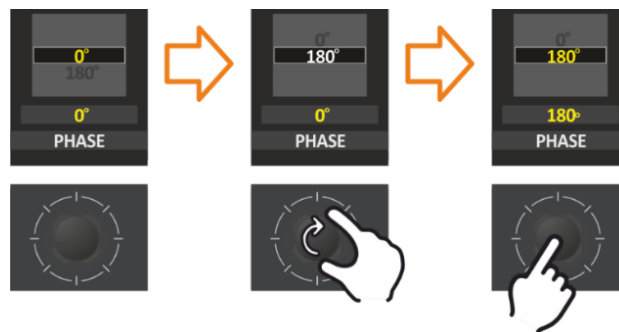
COLOR ON defines the colour transmitted to the selected DMX device when MIC 1 is ON. The selectable values are taken from the colour palette supported by the installed firmware and connected lighting device. For an RGB-compatible device, the available colours normally include combinations such as:

- RED
- GREEN
- BLUE
- YELLOW
- CYAN
- MAGENTA or PURPLE
- WHITE

Important: only colours supported by the selected device can be reproduced correctly. A fixture using a different DMX channel order or operating profile may display the wrong colour.

DIMMER ON:

DIMMER ON determines the brightness of the selected DMX device while MIC 1 is ON. The parameter can be adjusted from minimum to maximum intensity. **Range: 10% to 100%. Adjustment step: 1.0%**
Important: selecting a COLOR ON value is not sufficient if DIMMER ON is set to its minimum. The device may receive the correct colour command but remain dark because its brightness is set to zero.

**SET PREAMP WITH GAIN:**

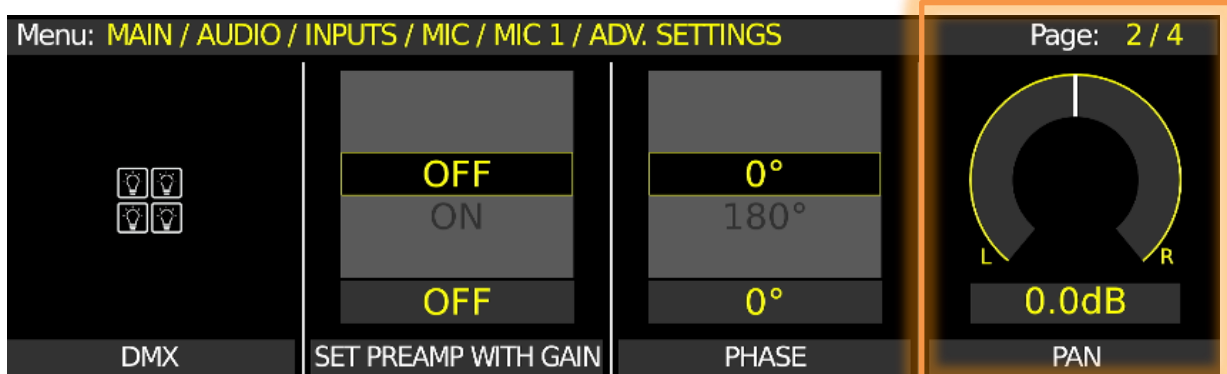
This function links the channel's **GAIN control** to the **analog microphone preamplifier**. When enabled, adjusting the Gain control also changes the preamp level, allowing the input signal to be optimized **before digital processing**. Correct adjustment improves the **signal-to-noise ratio** and prevents **clipping or distortion**.

PHASE:

This option was created to control the phase of the microphone by flipping the phase to prevent **PHASE CANCELLATION**. By rotating the knob of **the PHASE**, you can flip it **180°**.



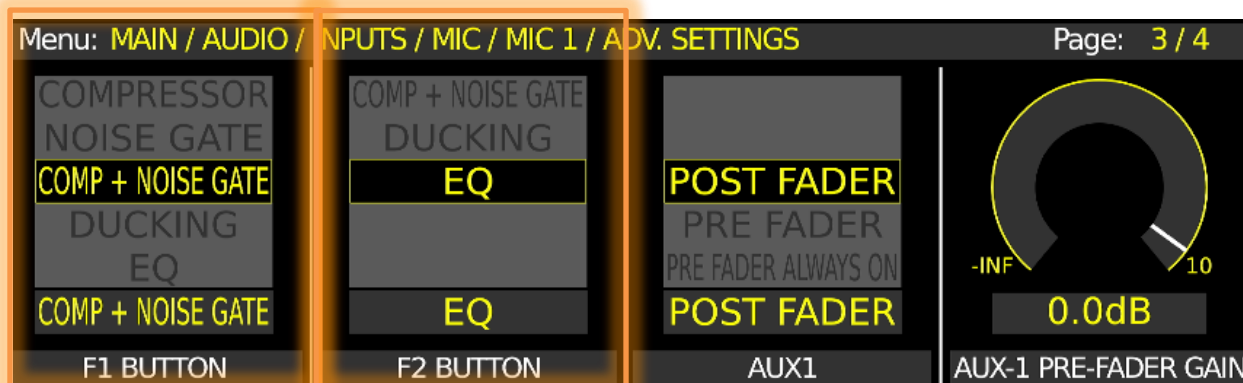
PAN:



This parameter changes the Input **PAN**. The parameter has a **1.0 dB** step.

N.B.: When you adjust the BAL/PAN parameter, you have a real-time change of the parameter value and a real-time perception of the BAL/PAN change. The Graphics will become white to indicate the parameter change, and they will become yellow again after the confirmation.

F1 & F2 buttons:



This feature gives the ability to change the action of the **F button**. For F1 & F2 to operate, they must be set to **UNLOCKED** mode. The F button can be assigned to:

- **NOISE GATE**: Assigns the Noise Gate function to the selected F1 or F2 button. When enabled, the Noise Gate attenuates the channel signal whenever its level falls below the configured threshold.
- **COMPRESSOR+NOISE GATE**: Assigns both the Compressor and Noise Gate functions to the selected F1 or F2 button. When enabled, the Noise Gate attenuates low-level background noise, while the Compressor controls signals exceeding the configured threshold.
- **COMPRESSOR**: Assigns the Compressor function to the selected F1 or F2 button. When enabled, the Compressor reduces the level of signals exceeding the configured threshold, according to the selected compression parameters.
- **NONE**: no action.
- **TB**: Assigns the Talkback function to the selected F1 or F2 button. When the button is pressed, the operator's microphone signal is routed to the configured studio communication output, allowing private communication with presenters, guests or other studio personnel. By pressing the F button can talk privately with the studio in Talkback (TB).

It can be set to enable/disable the (COMPRESSOR, DUCKING, and EQ) on this source.

AUX1, AUX2:



AUX-1 and AUX-2 are two independent **auxiliary audio buses** used to route the MIC 1 signal to destinations other than the main PGM or SUB outputs. They can be used for applications such as **headphone monitoring, studio communication, recording, streaming or external audio processing**.

The operating mode determines whether the signal sent to the AUX bus is affected by the channel fader and by the **ON/START status**. There are some different mode that could be set:

- **POST FADER:** The signal sent to the AUX bus follows the position of the MIC 1 channel fader. Increasing or decreasing the fader level produces the same level variation on AUX-1 or AUX-2. When the channel is switched off, the auxiliary signal is also muted. This mode is suitable when the auxiliary output must **follow the main channel mix**.
- **PRE FADER:** The signal is taken before the channel fader. Therefore, moving the MIC 1 fader does not change the level sent to the selected AUX bus. The channel's **ON/START status remains active**, so switching the channel off also disables the auxiliary signal. This mode is useful when an independent level is required for **monitoring, recording or communication**, while retaining control through the ON/START button.
- **PRE-FADER ALWAYS ON:** The signal is taken before the fader and remains available on the AUX bus regardless of both the **fader position and the ON/START status**. This mode provides a continuous microphone feed and is useful for permanent monitoring, talkback systems or dedicated recordings. It should be selected carefully because the microphone signal may remain present on the AUX output even when the channel is closed on the console.

AUX 1/2 PRE-FADER GAIN:



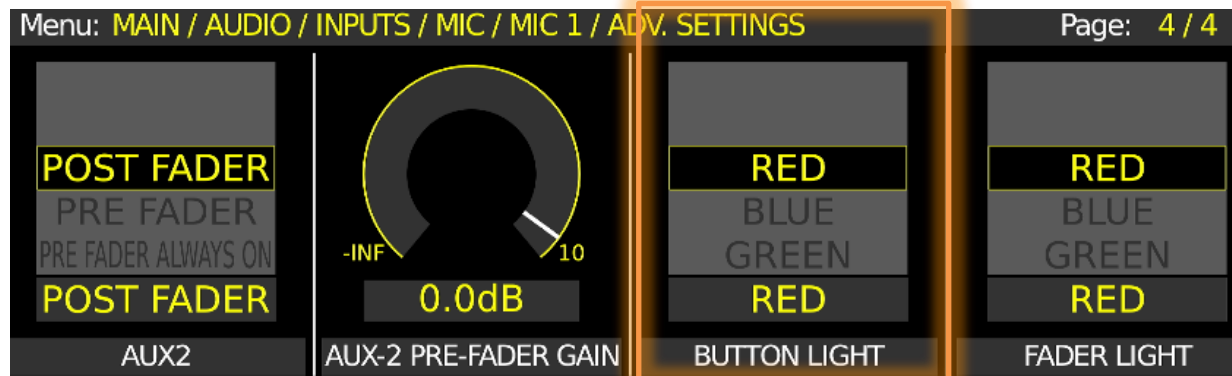
The **AUX 1 PRE-FADER GAIN** and **AUX 2 PRE-FADER GAIN** parameters, located inside the **ADV. SETTINGS** page of MIC 1, determine the level at which the microphone signal is sent to the corresponding auxiliary bus when a pre-fader operating mode is selected. They operate independently:

- AUX 1 PRE-FADER GAIN controls only the MIC 1 contribution to the AUX 1 bus.
- AUX 2 PRE-FADER GAIN controls only the MIC 1 contribution to the AUX 2 bus.

These controls make it possible to create two auxiliary mixes whose microphone levels are independent from the physical channel fader. **Range: -144.0 dB to +10.0 dB. Adjustment step: 1.0 dB**

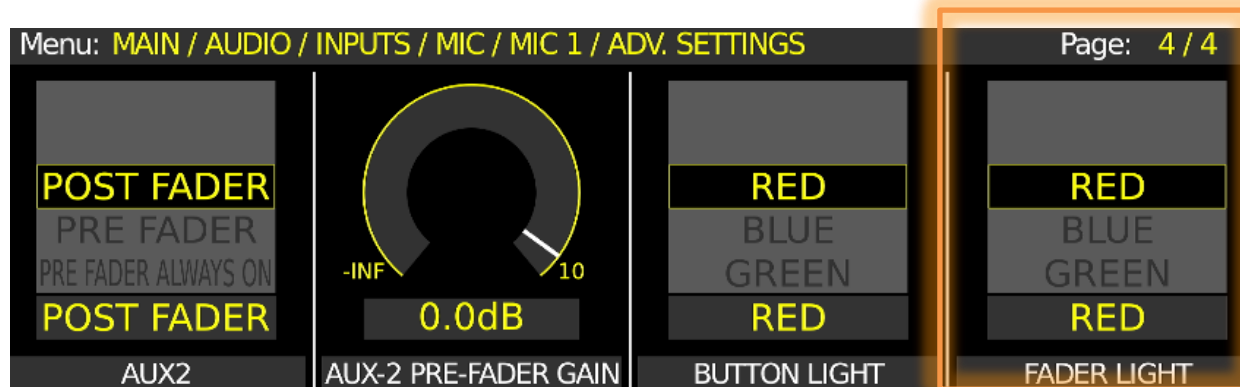
Firmware note: the precise minimum value, maximum value and adjustment step may depend on the installed firmware version. Always use the range displayed on the specific OXYGEN 3000 Plus being configured.

BUTTON-LIGHT:



Defines the **RGB illumination colour** used by the channel buttons when MIC 1 is assigned to a physical mixer channel. The selected colour provides immediate visual identification of the microphone source and helps distinguish it from other source types, such as telephone, VoIP or line inputs.

FADER BAR LIGHT:



Defines the **RGB colour of the LED bar located beside the channel fader** when MIC 1 is selected. The LED bar provides a visual indication of the **fader position and attenuation level**, while the assigned colour identifies the source type

MIC AUTOMIX

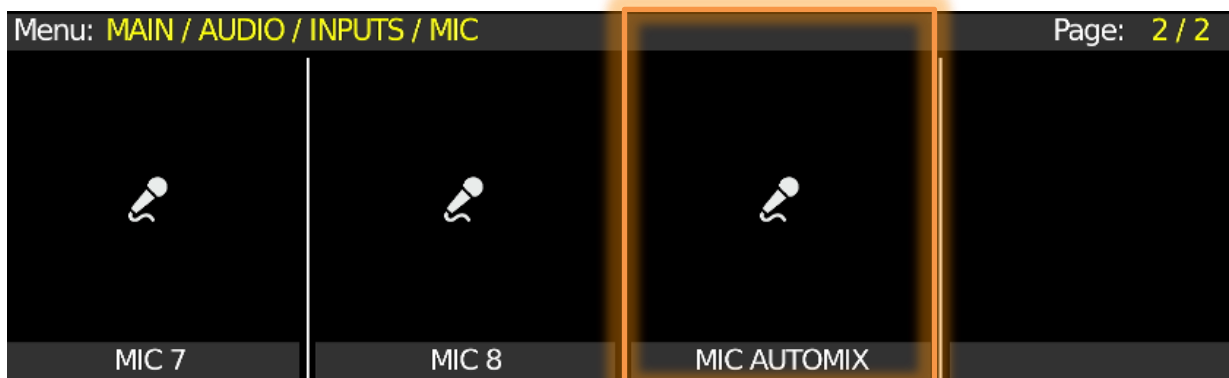
An automatic microphone mixer automatically reduces the strength of a microphone's audio signal when it is not being used. Automixers reduce extraneous noise picked up when several microphones operate simultaneously.

In the context of sound reinforcement systems, an automixer can be particularly useful for managing spontaneous discussions or panel discussions where multiple participants may be speaking at the same time or interjecting with sudden comments. By using an automixer, the operator can ensure that each participant's voice is heard clearly and that no words or phrases are lost due to **UPCUT mistakes** or lapses of attention.

UPCUT: Chopping off the beginning of the audio or video of a shot or video story. This happens when the editor or technical director doesn't cut to a new audio or video source quickly enough. Opposite of **downcut**.

Overall, while a skilled mixing engineer can greatly enhance the performance of a sound reinforcement system, the use of an automixer can provide an additional layer of support to ensure that all participants are heard clearly and that the audio quality remains consistent throughout the discussion.

To set up the **AUTOMIX**, navigate to **MAIN / AUDIO / INPUTS / MIC / MIC AUTOMIX**

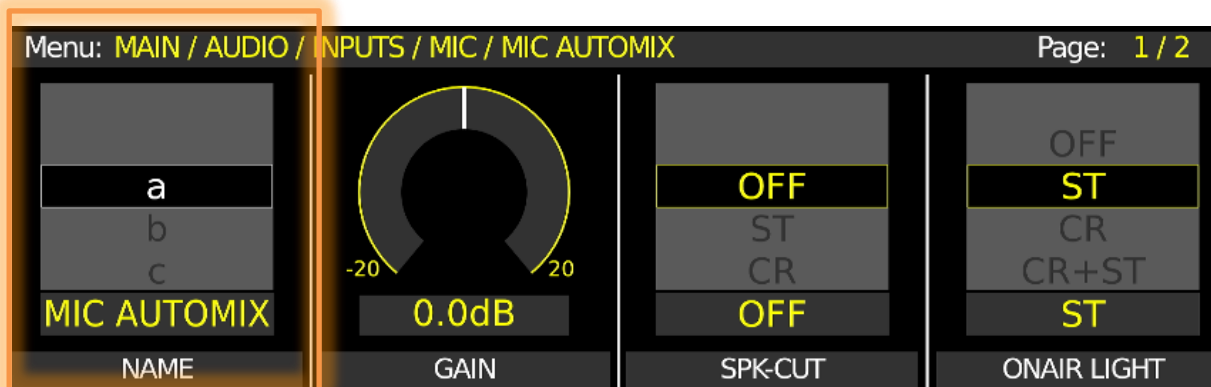


ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

There you can modify/set different options: **NAME, GAIN, SPK-CUT, ON AIR LIGHT, PRIVATE MIC, ON MODULE, ADVANCED SETTINGS.**

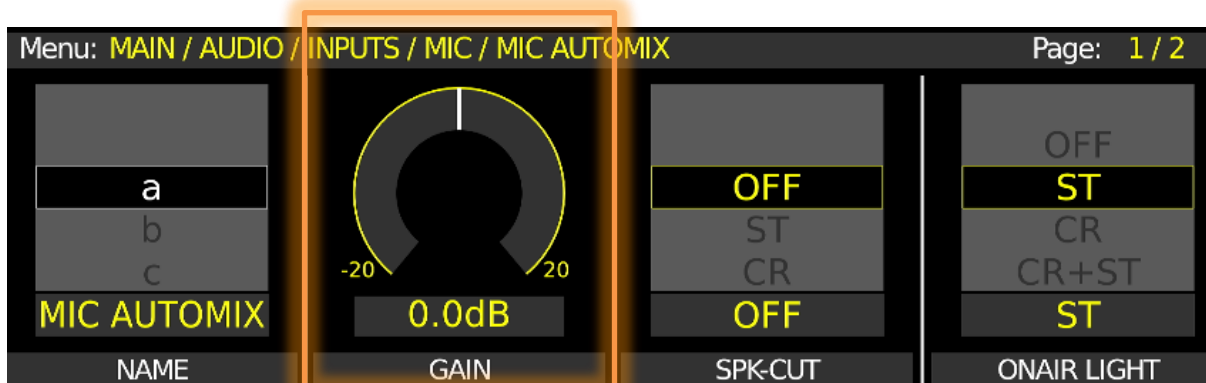
INPUTS SETTING (MIC AUTOMIX)

NAME:



Assigns a customized identification label to the MIC AUTOMIX source. The selected name is shown on the channel display and in the input-selection menus, allowing the operator to identify the source quickly. The label can contain up to **8 characters**. After entering the required name, press **ESC** to complete the editing procedure.

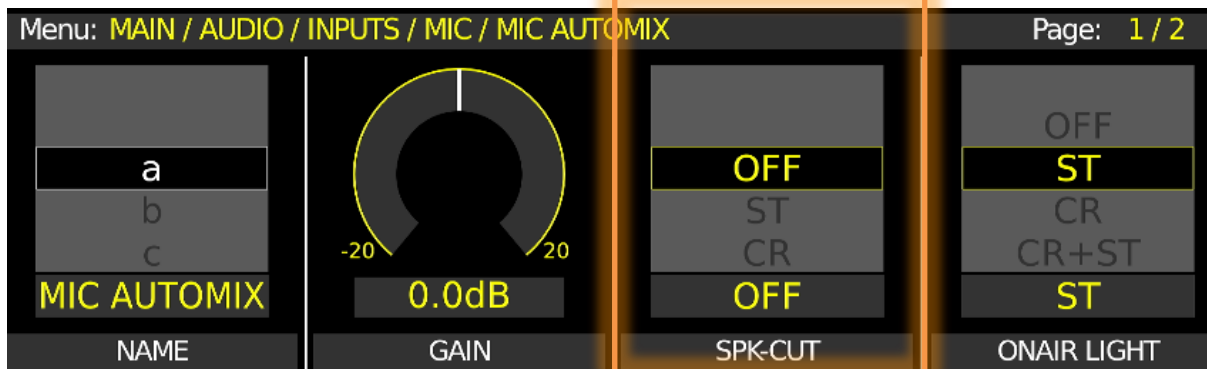
GAIN:



Adjusts the overall level of the MIC AUTOMIX signal before it is routed to the selected channel and output buses. This parameter affects the complete automatic microphone mix rather than the gain of each individual microphone. Correct adjustment provides a suitable output level while preserving sufficient headroom. Excessive gain may cause clipping or make background noise more noticeable.

Adjustment range: -20 dB to +20 dB. Step: 0.1 dB.

SPK CUT:

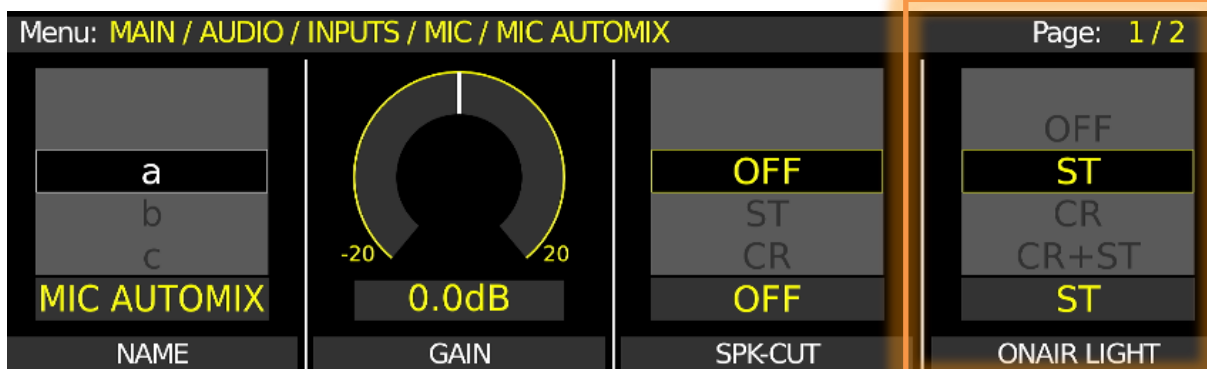


Defines which loudspeaker system is automatically muted when the MIC AUTOMIX source is opened. This prevents the mixed microphone signal from capturing the loudspeaker output and generating **acoustic feedback**.

- **OFF**: opening MIC AUTOMIX does not mute any loudspeaker output.
- **CR**: mutes the **Control Room loudspeakers**.
- **ST**: mutes the **Studio loudspeakers**.
- **CR+ST**: mutes both the Control Room and Studio loudspeakers.

SPK CUT affects only the selected loudspeaker outputs and does not mute programme, headphone or recording buses.

ON AIR LIGHT:

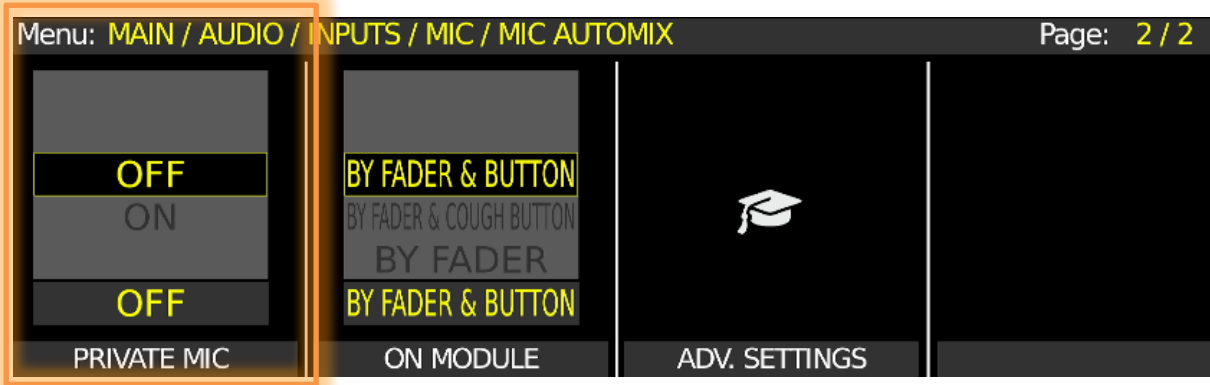


Associates the MIC AUTOMIX source with the external **On-Air signalling system**. When the source is opened, the console activates the corresponding programmable GPO command.

- **OFF**: MIC AUTOMIX does not activate an On-Air Light command.
- **CR**: activates the GPO assigned to the **Control Room On-Air Light**.
- **ST**: activates the GPO assigned to the **Studio On-Air Light**.
- **CR+ST**: activates both On-Air Light commands.

The corresponding GPO outputs must be configured in: [MENU / GENERAL SET. / GPIO / GPO](#)

PRIVATE MIC:

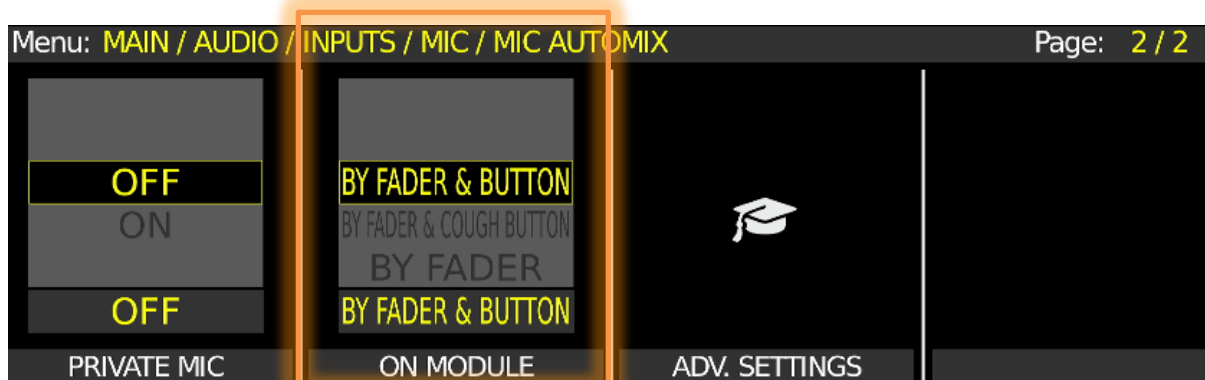


Defines whether the MIC AUTOMIX signal can be used for **private communication with a telephone caller**. When enabled, pressing PFL on the corresponding telephone or Telco channel routes the automatic microphone mix to the caller without sending the conversation to the main outputs or loudspeakers.

- **OFF:** MIC AUTOMIX is excluded from the private communication circuit.
- **ON:** MIC AUTOMIX can communicate with the caller when PFL is enabled on the telephone channel.

Because this function uses the complete Automix signal, all microphones currently contributing to the automatic mix may be heard by the caller.

ON MODULE:



Defines how the MIC AUTOMIX source is switched on or off when controlled through the associated module or TalkBox controls.

- **FADER & BUTTON:** the source status can be controlled through both the channel fader and the ON/START or TalkBox button.
- **BY FADER:** the source is switched on when the fader crosses the activation threshold and switched off when it moves below that threshold.
- **BY BUTTON:** the source status is controlled by the ON/START or TalkBox button. Moving the fader does not automatically change its ON/OFF status.
- **BY FADER & COUGH BUTTON:** the fader controls normal source activation, while the **Cough button temporarily mutes the microphone mix**. Releasing the button restores the previous operating condition.
- **ALWAYS ON:** MIC AUTOMIX remains permanently enabled on the module. **Important:** The **ALWAYS ON** option should be used carefully because the microphone mix may remain active independently of the normal console controls.
- **ALWAYS OFF:** MIC AUTOMIX remains permanently disabled on the module.

ADVANCED SETTINGS:

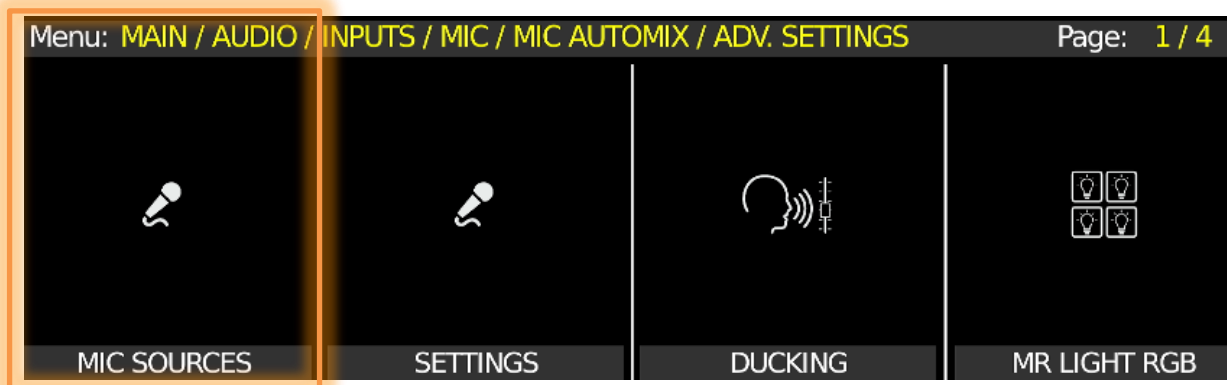


Opens the additional configuration section for the MIC AUTOMIX source. This menu provides access to the source's detailed **audio processing, routing and surface-control parameters**, allowing the automatic mix to be adapted to the required studio workflow. The ADVANCED SETTINGS available in MIC AUTOMIX are essentially identical to those previously described for MIC, with the addition of the MIC SOURCES, SETTINGS options, F1 MODE and F2 MODE.

ATTENTION: The remaining **ADVANCED SETTINGS** are identical to those previously described for **MIC 1–10**. For a detailed explanation of each function, refer to the corresponding section presented earlier in this manual. The settings not described here are **DUCKING, MR LIGHT RGB, DMX, PAN, BUTTON LIGHT, and FADER BAR LIGHT**.

-MIC SOURCES:

Determines which local and Dante microphone sources contribute to the combined **MIC AUTOMIX signal** and defines how each source is included in the automatic mix.



The available sources include:

- **MIC-1 to MIC-5:** local microphone inputs available with one Audio I/O board.
- **MIC-1 to MIC-10:** local microphone inputs available with two Audio I/O boards.
- **DANTE MONO 1 MODE to DANTE MONO 6 MODE:** mono audio sources received through the optional **Dante Audio-over-IP interface**.

Each source provides the following operating options:

- **OFF:** the source is permanently excluded from MIC AUTOMIX, regardless of its operating status on the console.
- **ON:** the source is permanently included in MIC AUTOMIX and processed by the automatic mixing algorithm.

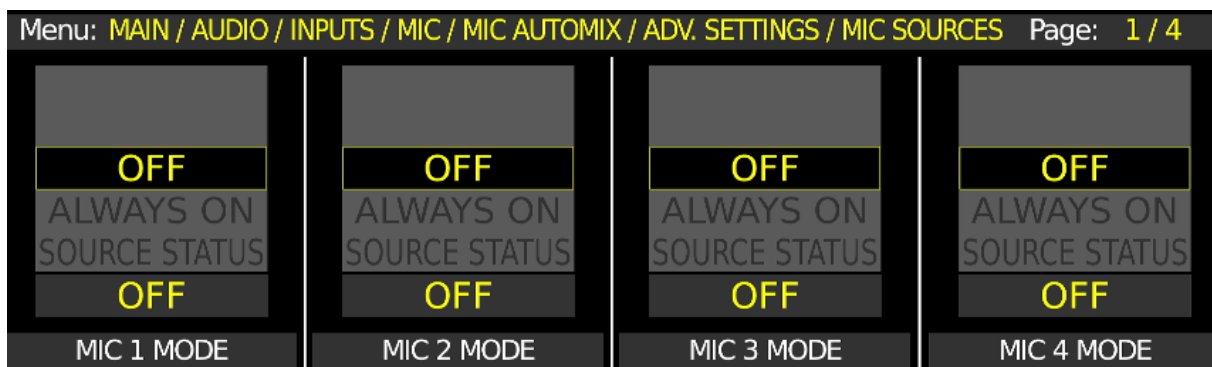
- **SOURCE STATUS:** the source's participation in MIC AUTOMIX follows its current **ON/OFF status**. When the source is ON, it is included in the automatic mix; when it is OFF, it is automatically excluded.

DANTE MONO 1 MODE – DANTE MONO 6 MODE:

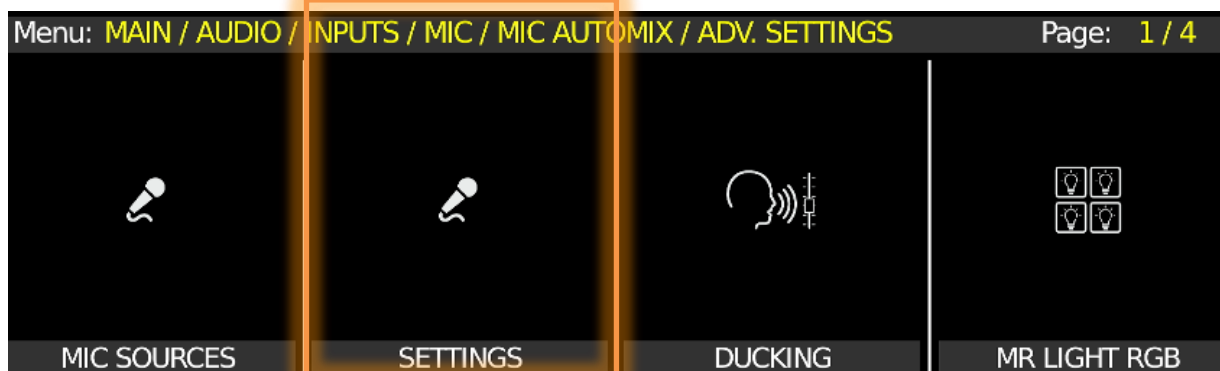
Configures the participation of each Dante mono source in MIC AUTOMIX. These sources can carry remote microphones, stage boxes, external microphone preamplifiers or other mono signals transported over the **Dante network**.

The Dante source must first be correctly configured and routed to the Oxygen 3000 Plus using Dante Controller. It must also be available as a **mono input source** in the console configuration.

Using **SOURCE STATUS** allows each Dante mono input to enter or leave the automatic mix dynamically according to its current console status. Selecting **ON** includes it continuously, while **OFF** excludes it completely. Excluding a local or Dante microphone from MIC AUTOMIX does not disable its individual input. It only prevents that source from contributing to the **combined Automix output**.

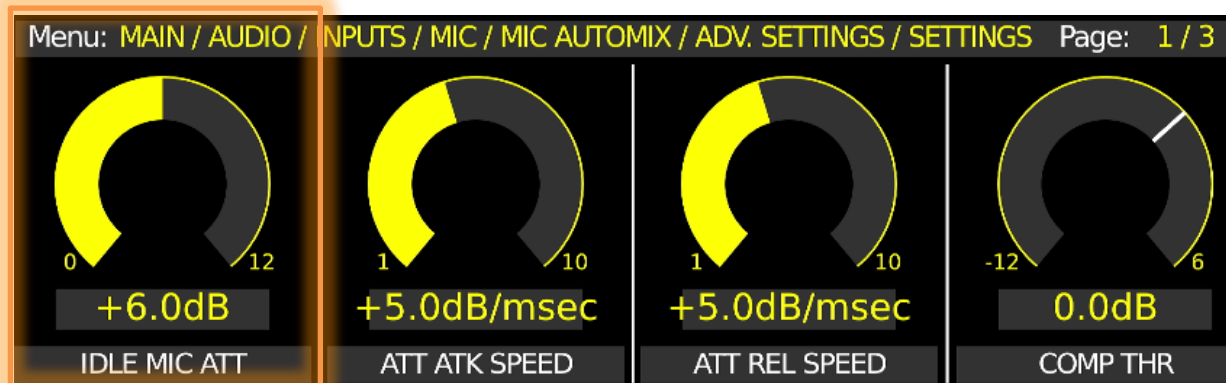


-SETTINGS:



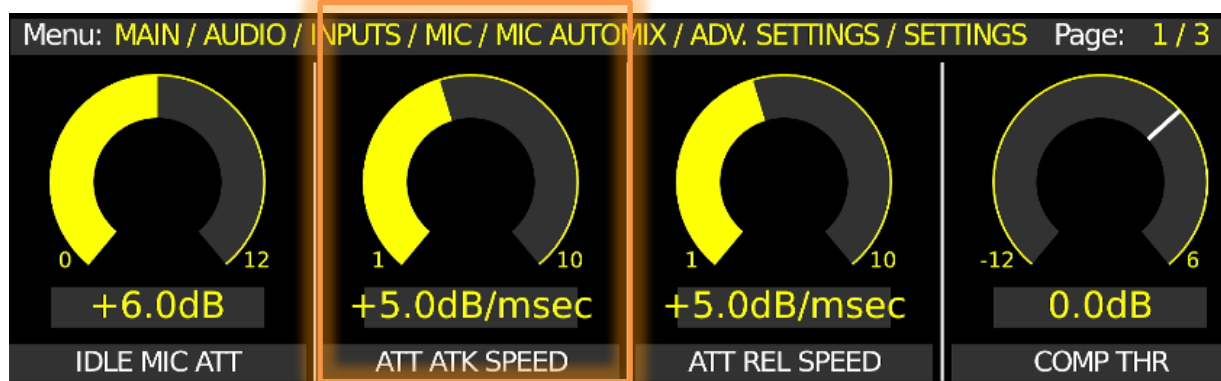
Provides access to the operating parameters of the Automix algorithm. These controls determine how inactive microphones are attenuated, how quickly the system responds to speech and how the final combined signal is levelled.

IDLE MIC ATTENUATION:



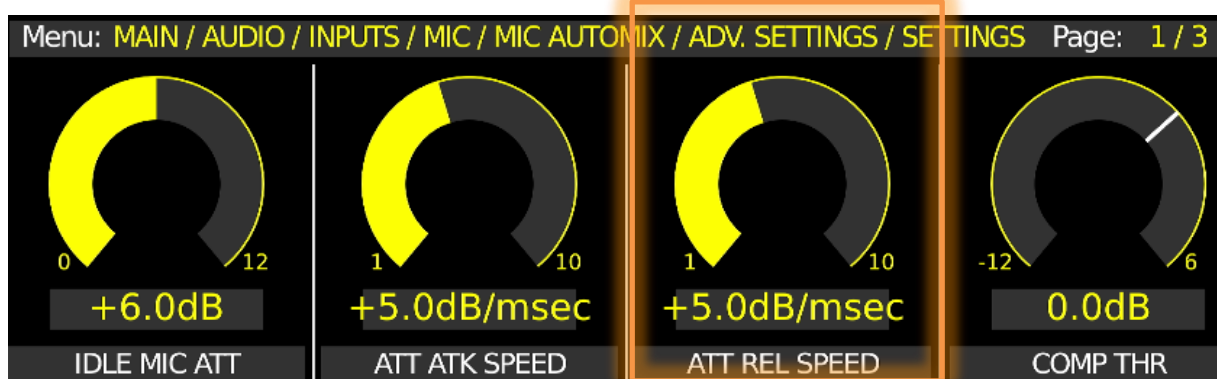
Defines the amount of level reduction applied to a microphone when the Automix algorithm identifies it as inactive. This reduces background noise and room reverberation without necessarily muting the microphone completely. A greater attenuation produces stronger isolation between microphones. Excessive attenuation may make transitions between speakers sound abrupt. **Range: 0 dB to 12.0 dB. Step: 1.0 dB.**

ATTENUATION ATTACK SPEED:



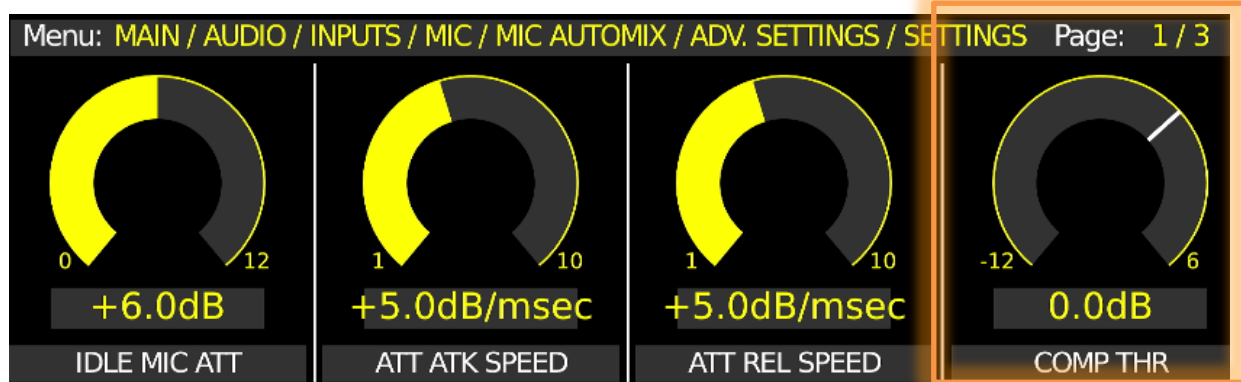
Defines how quickly an inactive microphone is reduced to the configured Idle Mic Attenuation level. A fast Attack reduces unwanted noise rapidly, while a slower setting provides a smoother transition. An excessively fast value may cause unnatural level changes during short pauses in speech. **Range: 1.0 dB/msec to 10.0 dB/msec. Step: 1.0 dB/msec.**

ATTENUATION RELEASE SPEED:



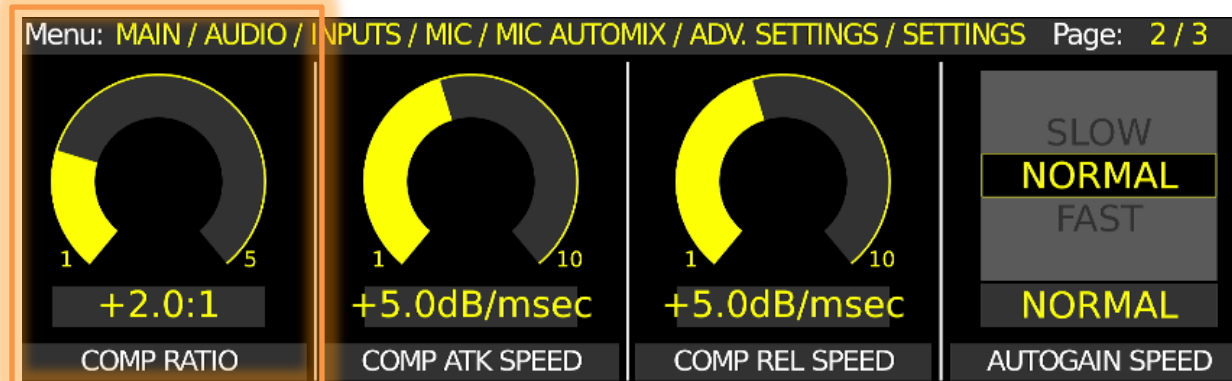
Defines how quickly an attenuated microphone returns to its normal level when speech is detected. A fast Release provides an immediate response, while a slower Release produces a smoother opening. If the release is too slow, the beginning of a sentence may be partially attenuated. **Range: 1.0 dB/msec to 10.0 dB/msec. Step: 1.0 dB/msec.**

COMPRESSION THRESHOLD:



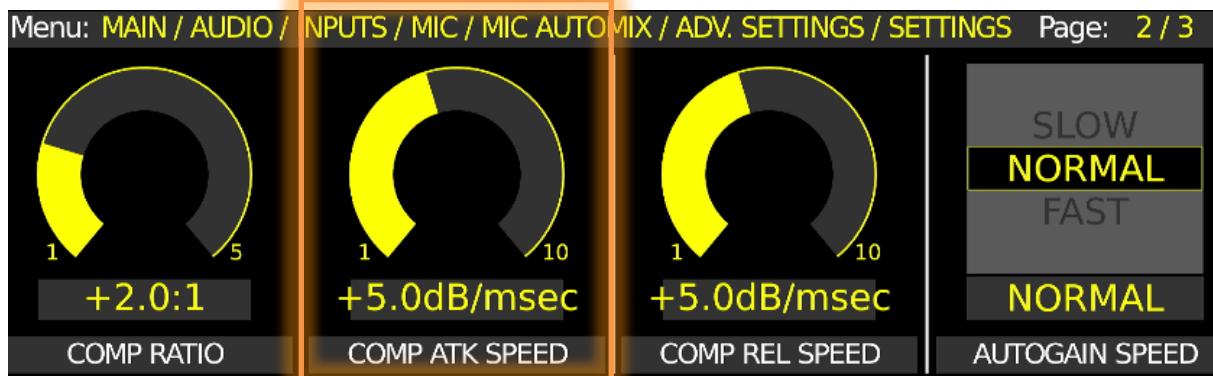
Sets the level above which the compressor begins processing the final MIC AUTOMIX signal. Signals below the threshold remain unchanged, while signals above it are reduced according to the selected Compression Ratio. A lower threshold applies compression to a larger portion of the combined signal. **Range: -12.0 dB to 6.0 dB. Step: 0.1 dB.**

COMPRESSION RATIO:



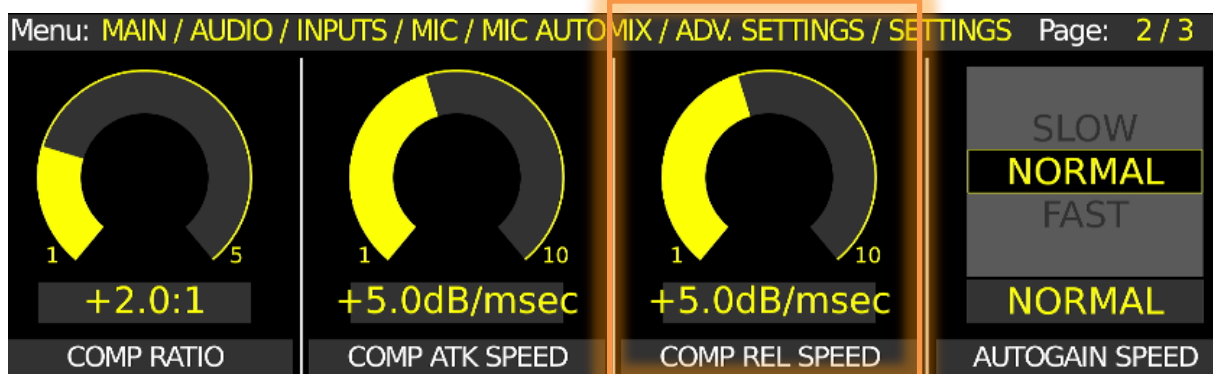
Defines the amount of gain reduction applied when the MIC AUTOMIX signal exceeds the Compression Threshold. For example, with a ratio of **4:1**, every 4 dB above the threshold produces only a 1 dB increase at the output. Higher values provide stronger control of level variations.

COMPRESSION ATTACK SPEED:



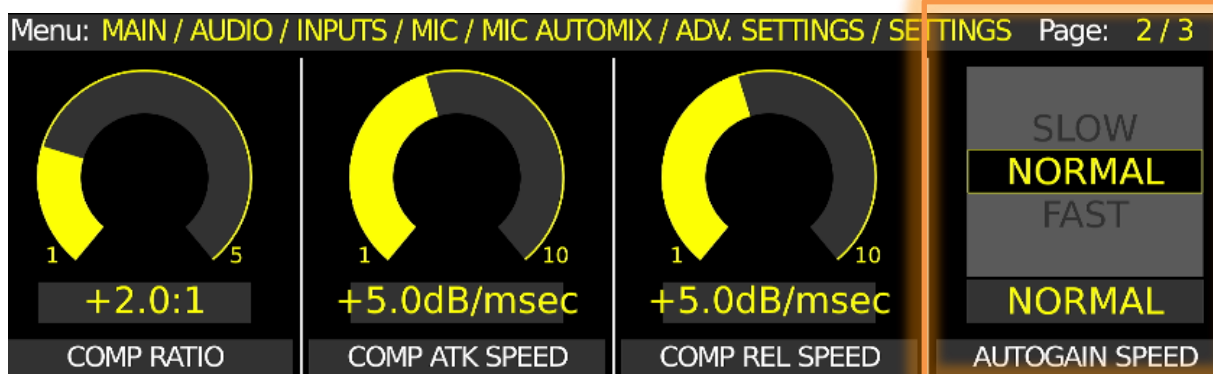
Defines how quickly the compressor reduces the level after the combined MIC AUTOMIX signal exceeds the threshold. A short Attack controls peaks rapidly, while a longer Attack preserves more of the natural initial transient of speech. **Range: 1.0 dB/msec to 10.0 dB/msec. Step: 1.0 dB/msec.**

COMPRESSION RELEASE SPEED:



Defines how quickly the compressor returns to its normal gain after the MIC AUTOMIX signal falls below the threshold. A longer Release produces smoother processing, while an excessively short Release may generate audible level fluctuations or pumping. **Range: 1.0 dB/msec to 10.0 dB/msec. Step: 1.0 dB/msec.**

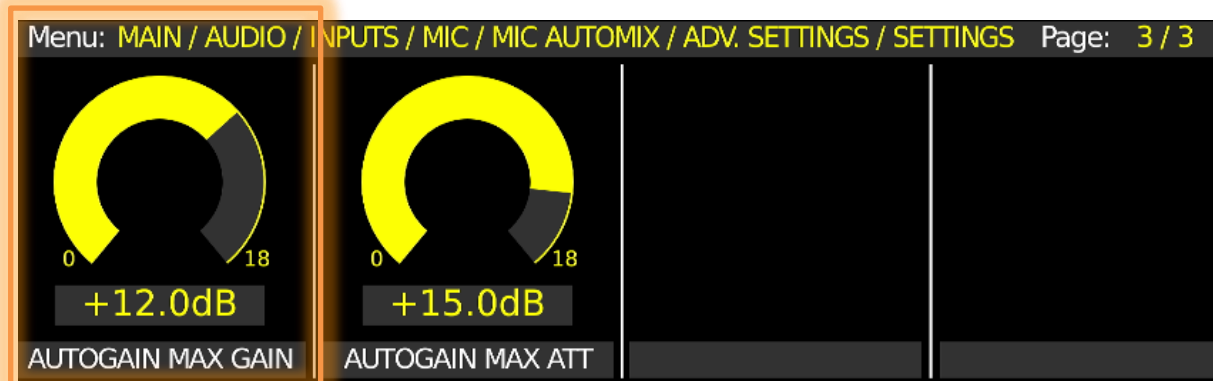
AUTOGAIN SPEED:



Defines how quickly the automatic gain-control section corrects level differences between the microphones included in Automix. A faster setting reacts rapidly to changing speakers, while a slower

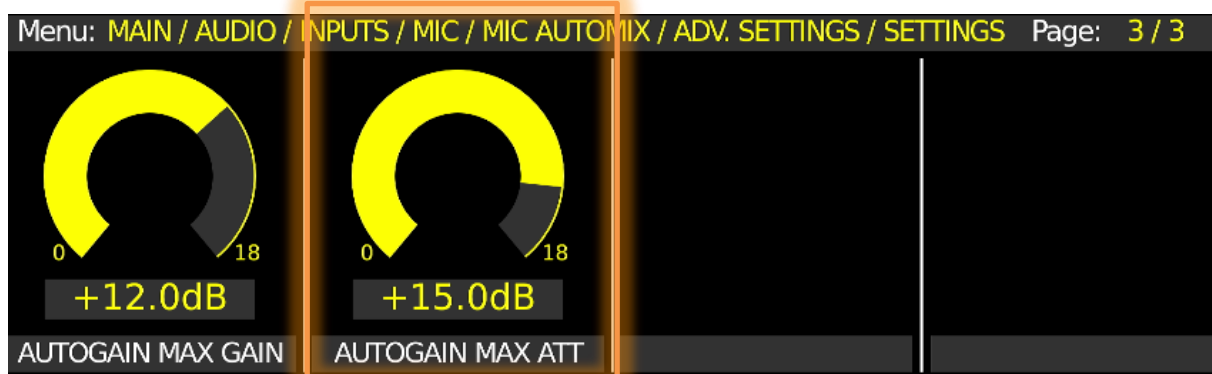
setting produces more gradual and natural adjustments. Excessively fast Autogain may make background noise or small level variations more noticeable.

AUTOGAIN MAX GAIN:



Sets the maximum positive gain that the Automix algorithm can apply to a microphone signal. It prevents low-level microphones from being amplified excessively. A high value may increase background noise or acoustic leakage. **Range: 0 dB to 18.0 dB. Step: 1.0 dB/msec.**

AUTOGAIN MAX ATTENUATION:



Sets the maximum amount by which the automatic gain-control section can reduce a microphone signal. It limits the downward correction applied when a microphone is significantly louder than the other selected sources. This parameter helps maintain a balanced mix while preventing excessive reduction of the wanted signal. **Range: 0 dB to 18.0 dB. Step: 1.0 dB/msec.**

-F1 MODE:

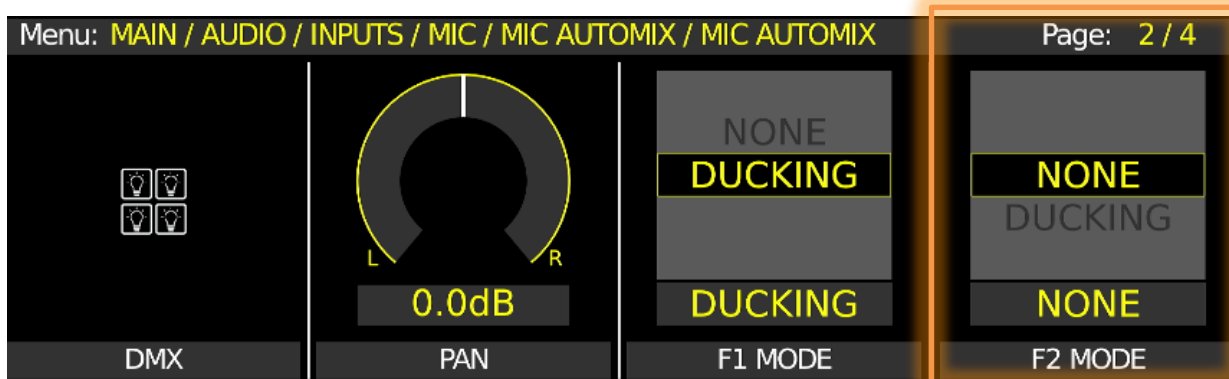


Defines the function assigned to the channel's **F1 button** when MIC AUTOMIX is loaded on that channel.

- **NONE:** no function is assigned to F1. Pressing the button produces no action for the MIC AUTOMIX source.
- **DUCKING:** assigns control of the MIC AUTOMIX **Ducking function** to F1. Pressing the button enables or disables the Ducking operation associated with the source, provided that the corresponding Ducking Status is configured as **UNLOCKED**. The button LED indicates the current operating status.

If the Ducking Status is set to **LOCKED ON** or **LOCKED OFF**, the assigned F1 button cannot change its state.

-F2 MODE:

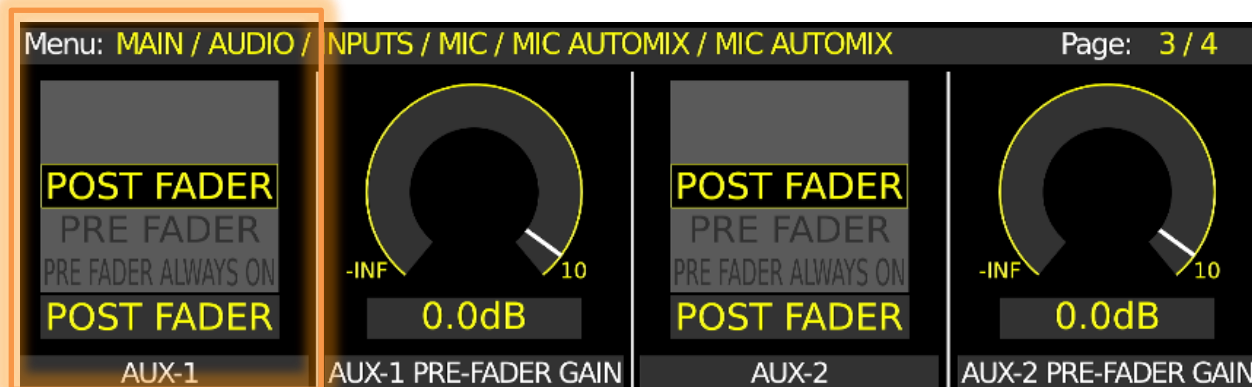


Defines the function assigned to the channel's **F2 button** when MIC AUTOMIX is loaded. It provides the same selectable options as F1 MODE.

- **NONE:** no function is assigned to F2.
- **DUCKING:** assigns control of the MIC AUTOMIX Ducking function to F2. The button can enable or disable Ducking only when the relevant Ducking Status is set to **UNLOCKED**.

Assigning DUCKING to both F1 and F2 is normally unnecessary because both buttons would control the same processing function. It is therefore recommended to assign **DUCKING to only one button** and leave the other set to **NONE**.

-AUX-1:

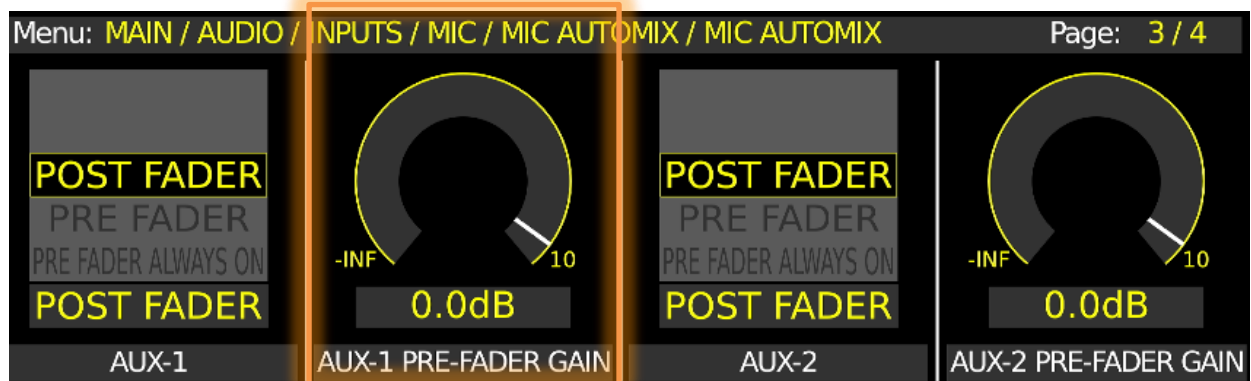


Defines how the combined MIC AUTOMIX signal is routed to the **AUX-1 bus**. The selected mode determines whether the auxiliary level follows the channel fader and ON/START status.

- **POST FADER:** the AUX-1 level follows the MIC AUTOMIX channel fader. Closing or reducing the fader produces the same change on AUX-1.
- **PRE FADER:** the AUX-1 signal is independent of the channel fader but remains controlled by the channel's ON/START status.

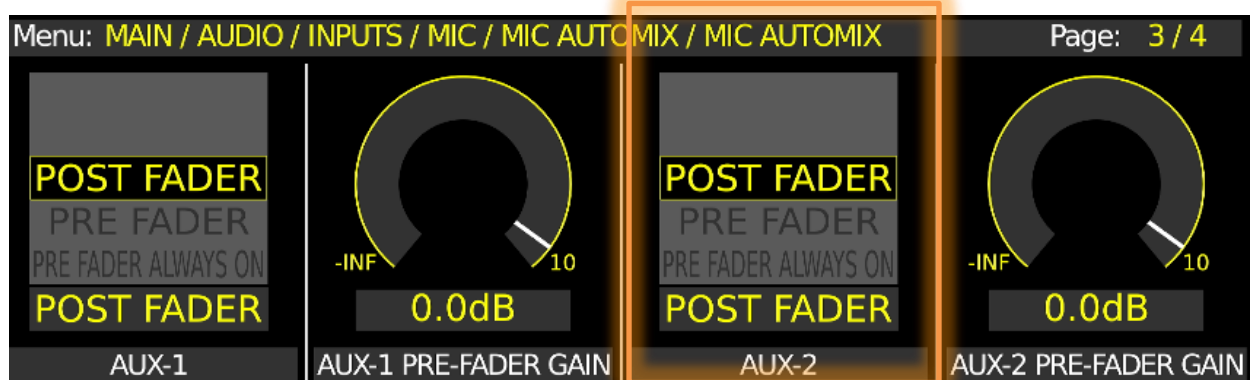
- **PRE-FADER ALWAYS ON:** the signal remains available on AUX-1 regardless of both the fader position and the ON/START status.

-AUX-1 PRE-FADER GAIN:



Adjusts the level of the combined MIC AUTOMIX signal sent to **AUX-1** when a **Pre-Fader mode** is selected. This provides an auxiliary level that is independent of the main channel fader and is useful for headphone monitoring, recording, communication or external processing. Excessive gain may cause **overload or clipping on the AUX-1 bus**.

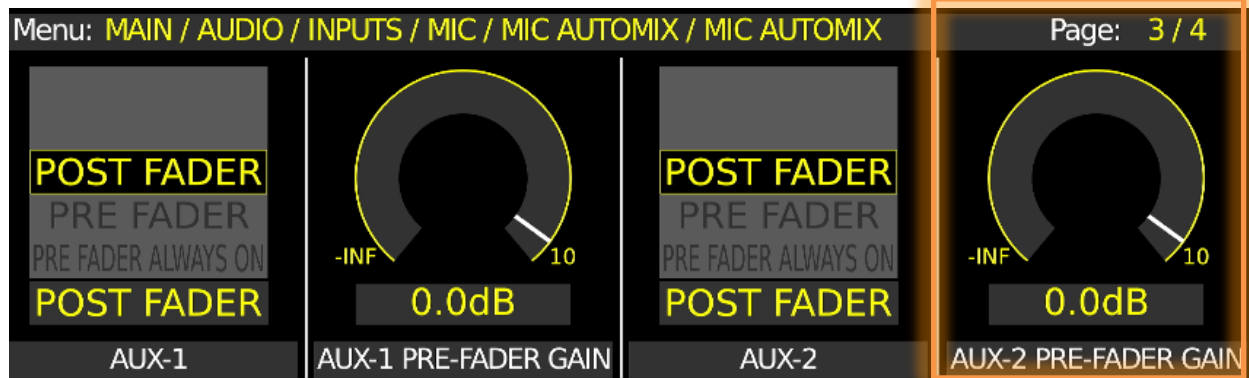
-AUX-2:



Defines how the combined MIC AUTOMIX signal is routed to the **AUX-2 bus**. Its operation is independent from AUX-1.

- **POST FADER:** the AUX-2 level follows the MIC AUTOMIX channel fader.
- **PRE FADER:** the AUX-2 signal is independent of the channel fader but follows the channel's ON/START status.
- **PRE-FADER ALWAYS ON:** the MIC AUTOMIX signal remains continuously available on AUX-2, independently of the fader and ON/START controls.

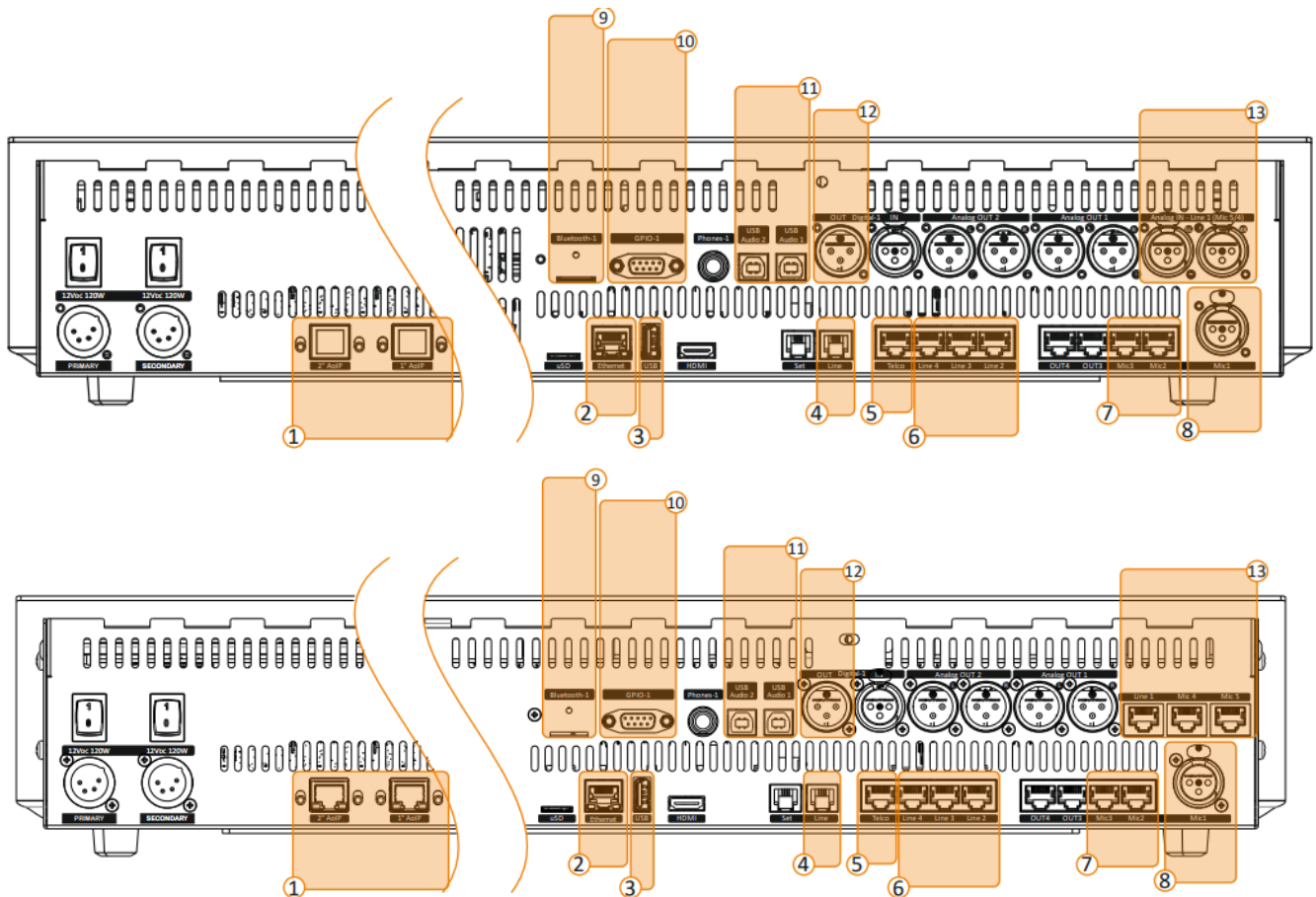
-AUX-2 PRE-FADER GAIN:



Adjusts the level of the combined MIC AUTOMIX signal sent to **AUX-2** when a **Pre-Fader mode** is selected. AUX-1 and AUX-2 Pre-Fader Gain are independent, allowing two different auxiliary mixes to be created from the same Automix source. Correct adjustment ensures an appropriate signal level while preventing **noise, overload and digital clipping**.

MONO:

A **MONO input** carries a single audio channel, unlike a stereo input, which uses two separate **Left and Right channels**. On the Oxygen 3000 Plus, each physical stereo input can be configured as two independent mono sources through the **INPUT MODE** settings. When **MONO mode** is selected, the Left and Right signals of the original stereo connection can be processed, routed and assigned separately. When the input is configured as **STEREO**, the related mono sources are not available. MONO inputs can be used for microphones connected through an external preamplifier, telephone hybrids, codecs, mono audio players or any other balanced line-level source. The connection is provided through a **female RJ45 connector** for balanced audio, with an input impedance of **50 kΩ**. Depending on the Oxygen 3000 Plus configuration, the **3 or 6 stereo inputs (L/R)** can therefore be converted into **6 or 12 independent mono inputs**, respectively.

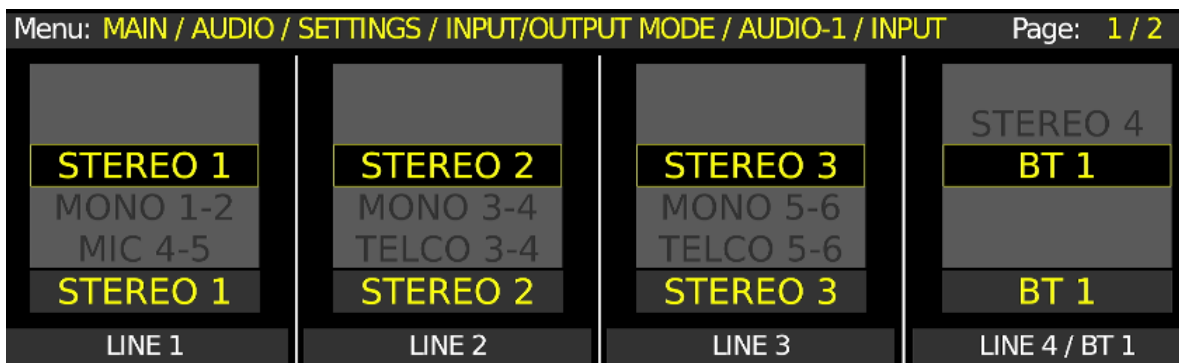


Important: The audio input connections for **MONO** are located on the **first Audio Board**, at positions **6,13**.

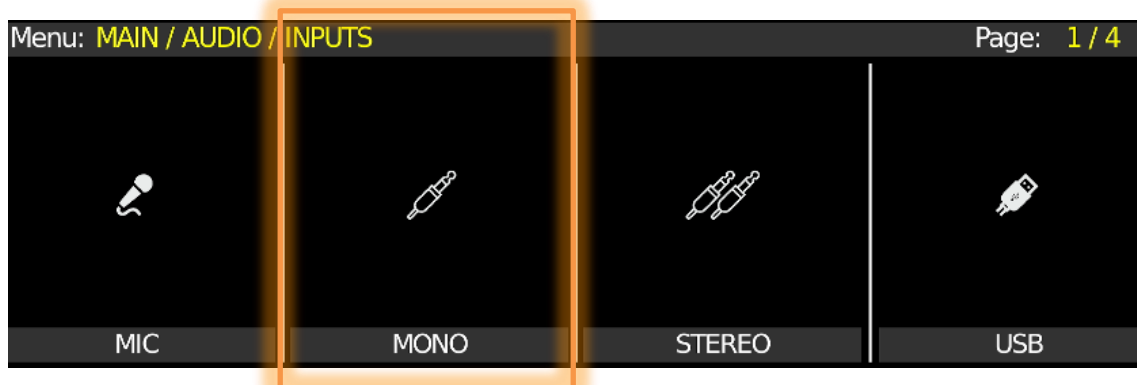
ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

To change the source type from **MONO** to **STEREO** or on contrary you need to follow these steps: -

MENU: **MAIN / AUDIO / SETTING / INPUT MODE / AUDIO-1 / INPUT**



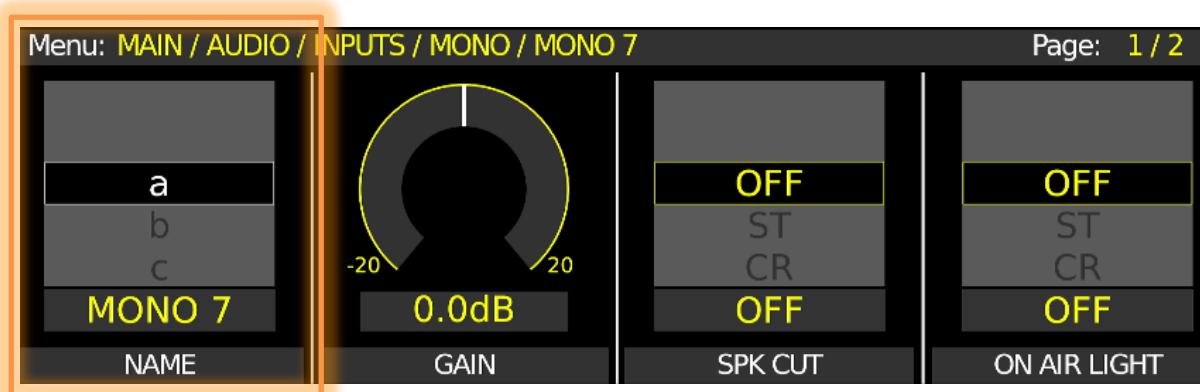
From this level of the menu, you choose which input **MONO/STEREO** source you like to customize.
MENU: *MAIN / AUDIO / INPUTS / (MONO or STEREO)*



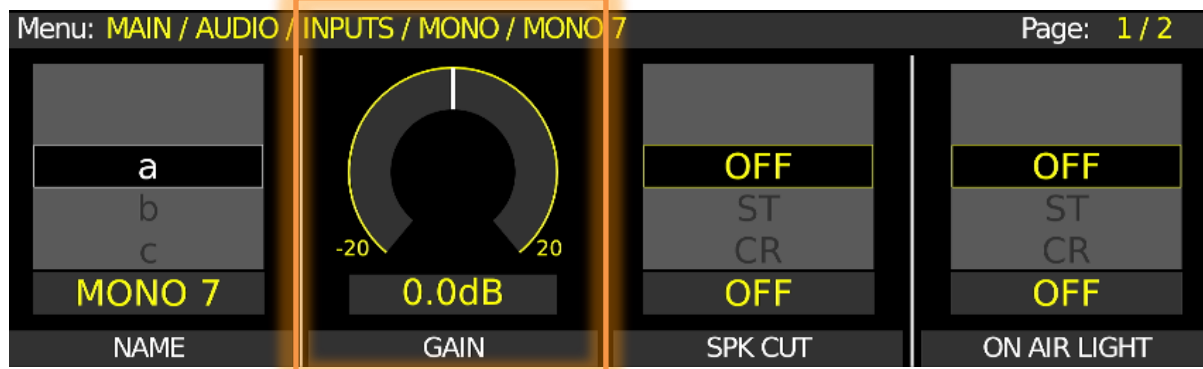
INPUTS SETTING (MONO)

Here we can find the source parameters (**MONO**). In this page you can modify/set different parameters of the MONO. The main modality are: **NAME, GAIN, SPK CUT, ON AIR LIGHT, PRIVATE MIC, TB MIC, ON MODULE ADV SETTINGS.**

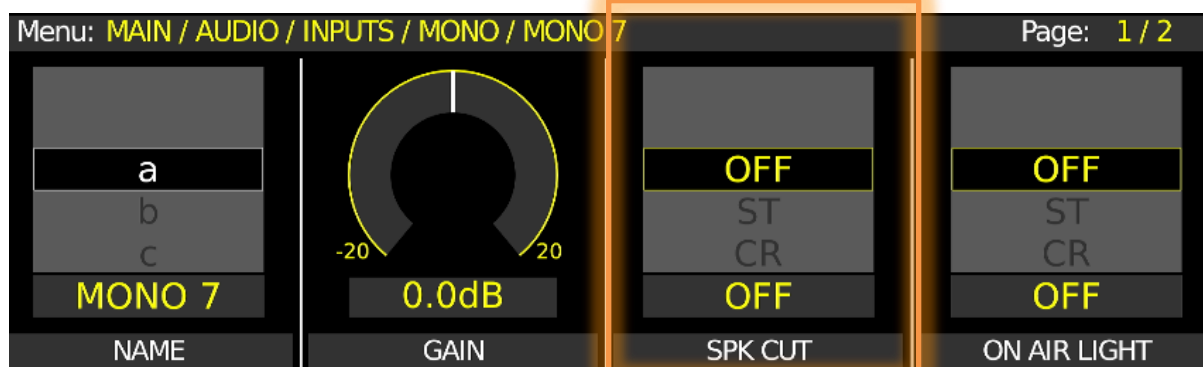
NAME:



Assigns a customized identification label to the selected MONO source. The name is displayed in the source-selection menus and on the channel display when the source is assigned to a mixer channel. The label can contain up to **8 characters**. Press **ESC** to complete the editing procedure.

GAIN:

Adjusts the digital input level of the selected MONO source. It allows the source level to be matched with the other inputs before routing and processing. Excessive positive gain may cause digital clipping, while excessive attenuation may produce an insufficient output level. **Adjustment range: -20 dB to +20 dB. Step: 0.1 dB.**

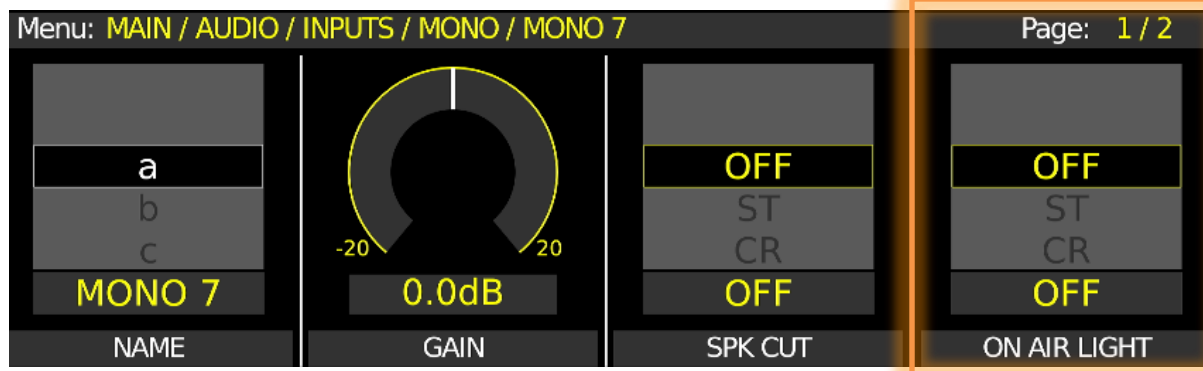
SPK CUT:

Defines which loudspeaker system is automatically muted when the selected MONO source is opened. This is useful when the MONO input carries a microphone signal that could generate **acoustic feedback**.

- **OFF:** opening the MONO source does not mute any loudspeaker output.
- **CR:** mutes the **Control Room loudspeakers**.
- **ST:** mutes the **Studio loudspeakers**.
- **CR+ST:** mutes both the Control Room and Studio loudspeakers.

SPK CUT affects only the selected loudspeaker outputs and does not mute programme, headphone or recording buses.

ON AIR LIGHT:

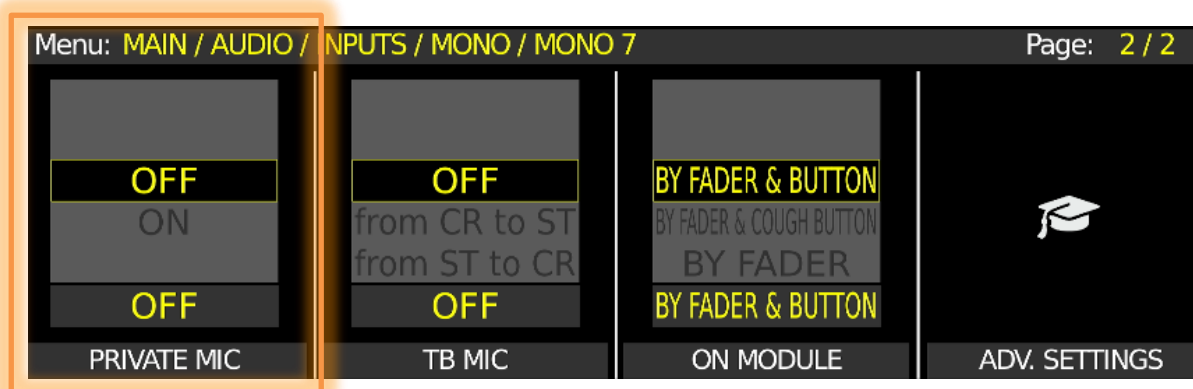


Associates the selected MONO source with the external **On-Air signalling system**. When the source is opened, the console activates the corresponding programmable GPO command.

- **OFF**: the MONO source does not activate an On-Air Light command.
- **CR**: activates the GPO assigned to the Control Room On-Air Light.
- **ST**: activates the GPO assigned to the Studio On-Air Light.
- **CR+ST**: activates both On-Air Light commands.

The corresponding GPO outputs must be configured in: [MENU / GENERAL SET. / GPIO / GPO](#)

PRIVATE MIC:

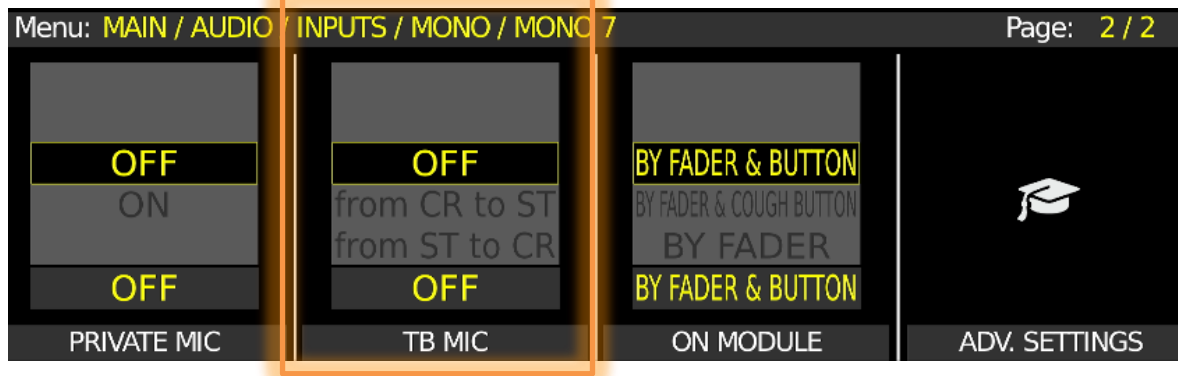


Defines whether the selected MONO source can be used for **private off-air communication with a telephone caller**. This option is useful when the MONO input carries a microphone signal from an external microphone preamplifier or communication system.

- **OFF**: the MONO source is excluded from the private communication circuit.
- **ON**: the MONO source is routed to the caller when PFL is enabled on the corresponding telephone or Telco channel.

The Private Mic signal is independent of the channel's **fader position and ON/START status**. It is not sent to the main programme outputs or loudspeakers.

TB MIC:

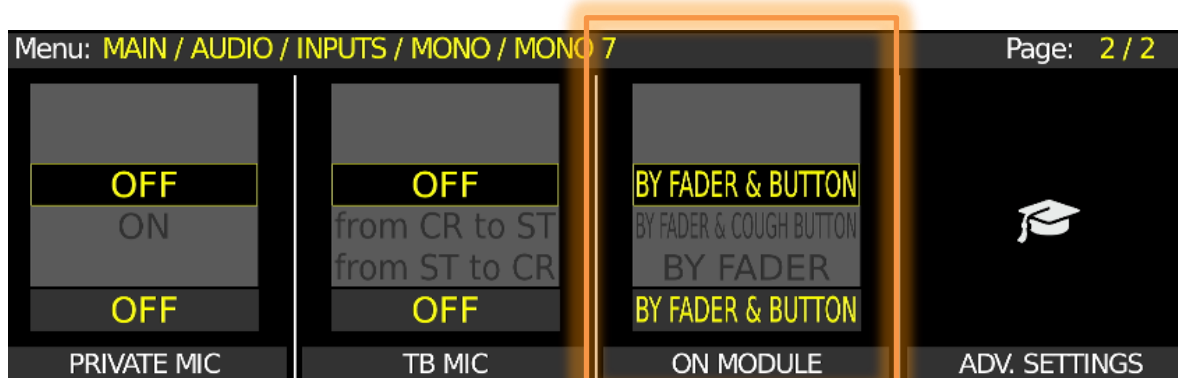


Configures the selected MONO source as a **Talkback microphone** and determines the direction of the internal communication.

- **OFF:** the source is disconnected from the Talkback system.
- **FROM CR TO ST:** the source is used for communication **from the Control Room to the Studio**.
- **FROM ST TO CR:** the source is used for communication **from the Studio to the Control Room**.

The Talkback signal is routed only to the configured monitoring destination and is normally excluded from the main On-Air buses.

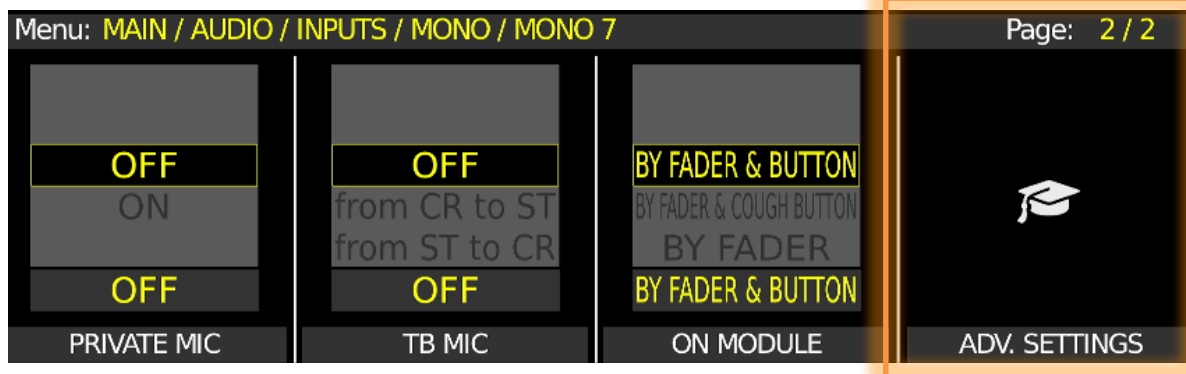
ON MODULE:



Defines how the selected MONO source is switched on or off when assigned to a mixer channel or associated control module.

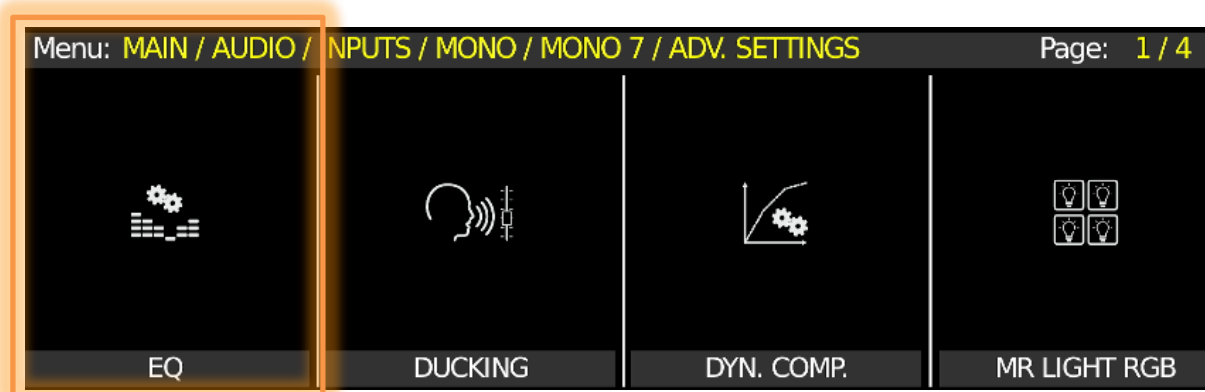
- **FADER & BUTTON:** the source status can be controlled through both the channel fader and the ON/START button.
- **BY FADER:** the source is switched on when the fader crosses the activation threshold and switched off when it moves below that threshold.
- **BY BUTTON:** the source status is controlled by the ON/START button. Moving the fader does not automatically change its ON/OFF status.
- **BY FADER & COUGH BUTTON:** the fader controls normal source activation, while the Cough button temporarily mutes the signal. Releasing the button restores the previous operating condition.
- **ALWAYS ON:** the MONO source remains permanently enabled. The **ALWAYS ON** option should be used carefully, particularly when the MONO input carries a microphone signal.
- **ALWAYS OFF:** the MONO source remains permanently disabled.

ADV. SETTINGS:



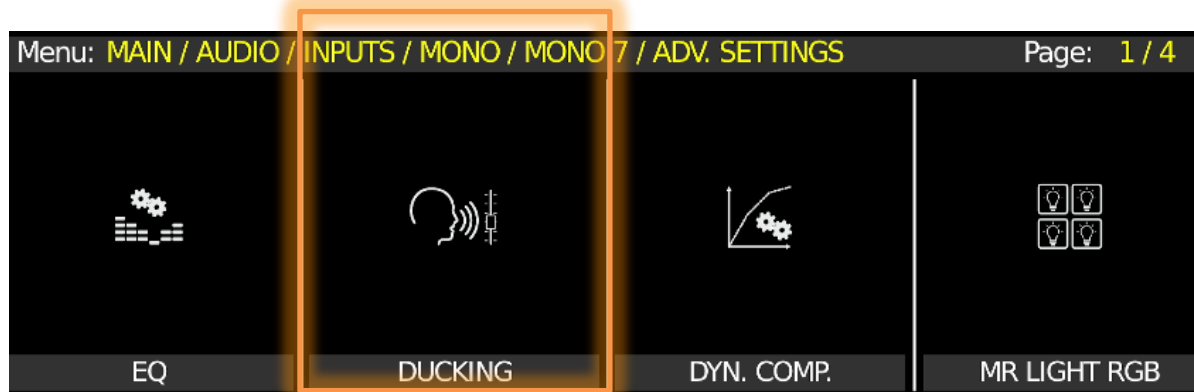
Opens the advanced configuration section for the selected MONO source. It provides access to detailed audio processing, routing and surface-control parameters. The option that you can set in ADVANCED SETTINGS are: **EQ, DYN. COMP, DUCKING, MR LIGHT RGB, PAN, DMX, F1 BUTTON, F2 BUTTON, AUX 1, AUX 1 PRE-FADER GAIN, AUX 2, AUX 2 PRE-FADER GAIN, BUTTON LIGHT, FADER BAR LIGHT.**

EQ:



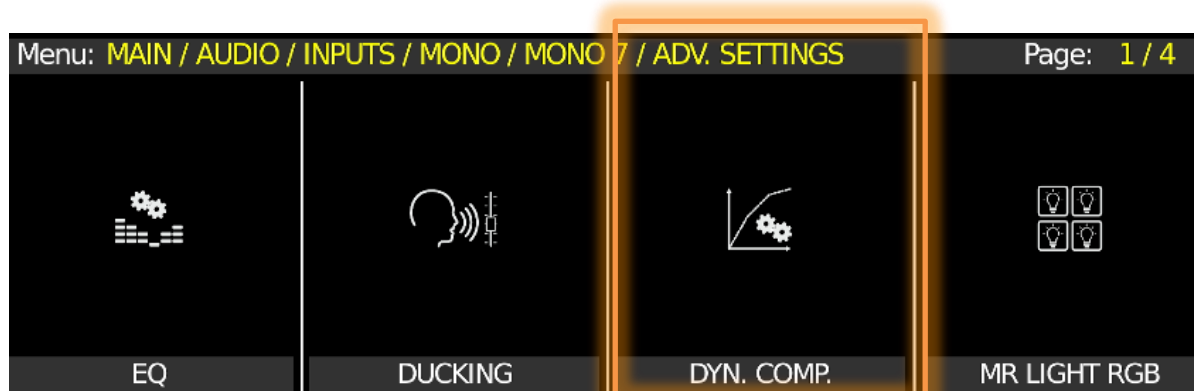
Opens the parametric Equalizer section for the selected MONO source. The EQ modifies the signal's frequency response, allowing unwanted frequencies to be attenuated and useful frequency ranges to be emphasized.

ATTENTION: A detailed description of this mode can be found in the section dedicated to the MIC input

DUCKING:

Configures the selected MONO source within the console's automatic **Master/Slave level-control system**. **SLAVE MODE** determines whether the MONO source is attenuated when another source operates as a Ducking Master. **MASTER MODE** determines whether the MONO source can control the attenuation of other sources configured as Slaves.

ATTENTION: A detailed description of this mode can be found in the section dedicated to the MIC input

DYN. COMP.:

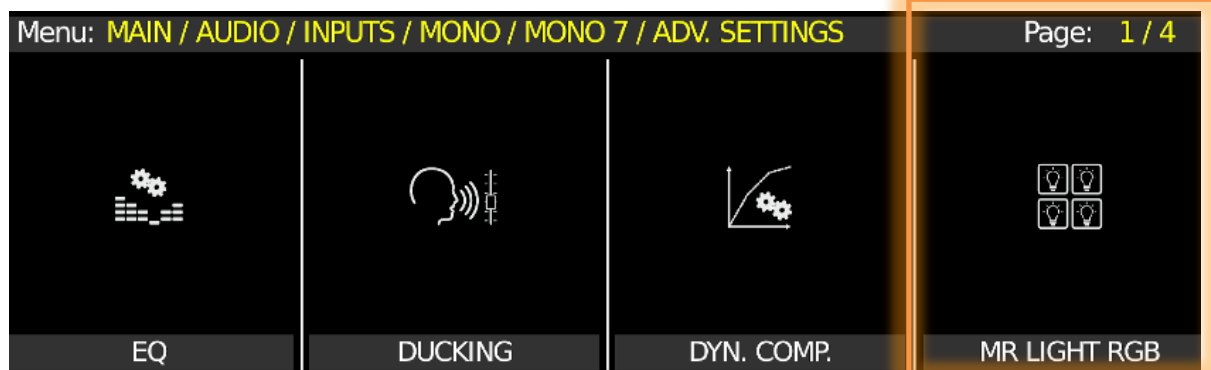
Opens the dynamic-processing section for the MONO source. It combines Compression, Limiting and Noise Gate/Expansion processing to control signal-level variations and reduce unwanted noise.

The main sections include:

- **DYNAMIC PRESET:** saves or recalls a predefined dynamic-processing configuration.
- **COMPRESSOR:** reduces signals that exceed the configured threshold.
- **NOISE GATE:** attenuates signals that fall below the configured expansion threshold.

ATTENTION: A detailed description of this mode can be found in the section dedicated to the MIC input

MR LIGHT RGB:

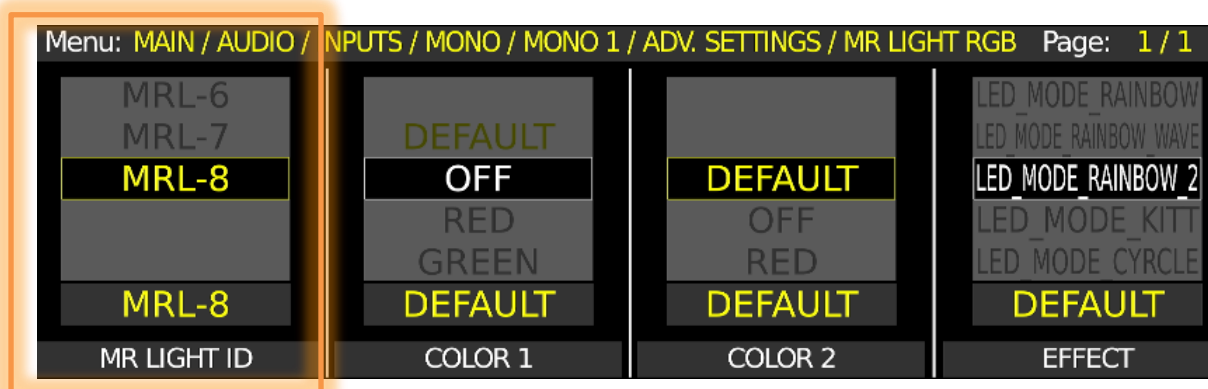


Associates the selected MONO source with one or more compatible **AxelTech MR. LIGHT RGB signalling lamps**. When the source changes to its configured active or On-Air state, the console can send a network command to control the lamp.

The available selections generally include:

- **OFF:** the source does not control an MR LIGHT RGB device.
- **ENABLED/ASSIGNED DEVICE:** associates the source with an MR LIGHT detected or configured on the network.

MR LIGHT ID:



MR LIGHT ID selects the MR Light RGB device that receives the command generated by MONO.

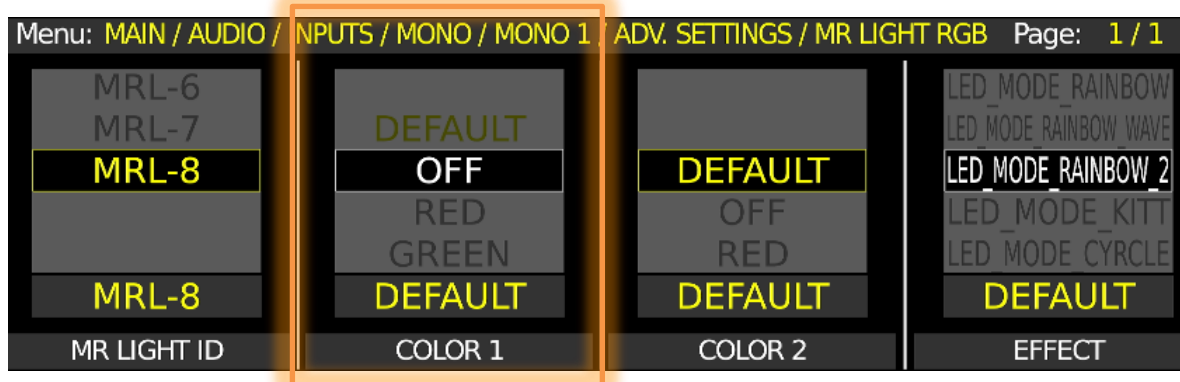
The selectable identifiers are:

- **NONE:** When **NONE** is selected, MONO does not directly control an MR Light RGB device. NONE may be selected when: MONO does not require individual visual signalling / The lamp is controlled through CR LIGHT or ST LIGHT logic / The MR Light RGB installation has not yet been completed
- **MRL-CR1 / MRL-CR2:** **MRL-CR1** and **MRL-CR2** identify the two device slots conventionally assigned to the Control Room. The slot name does not by itself determine the operating logic. The selected slot must also be correctly configured on the general MR LIGHT RGB page. When the device's CONTROL parameter is set to **ON**, MONO can control the corresponding lamp using its individual colour and effect settings. When CONTROL is set to **CR LIGHT**, the device instead follows the general Control Room light command.

- **MRL-ST1 / MRL-ST2:** **MRL-ST1** and **MRL-ST2** identify the two device slots conventionally assigned to the Studio. When the selected device is configured with **CONTROL = ON**, it can respond directly to the individual MONO configuration. When it is configured as **ST LIGHT**, it follows the general Studio light command instead.
- **MRL-1-8:** **MRL-1** through **MRL-8** are general-purpose device slots. They can be used for: Presenter or guest signalling / Additional On Air lamps / Production-area indicators

These identifiers correspond to the device slots available on the MR LIGHT RGB general configuration page. Each slot must be associated with the network parameters of the required lamp, including its IP address and, where provided, its MAC address and device name.

COLOR 1:



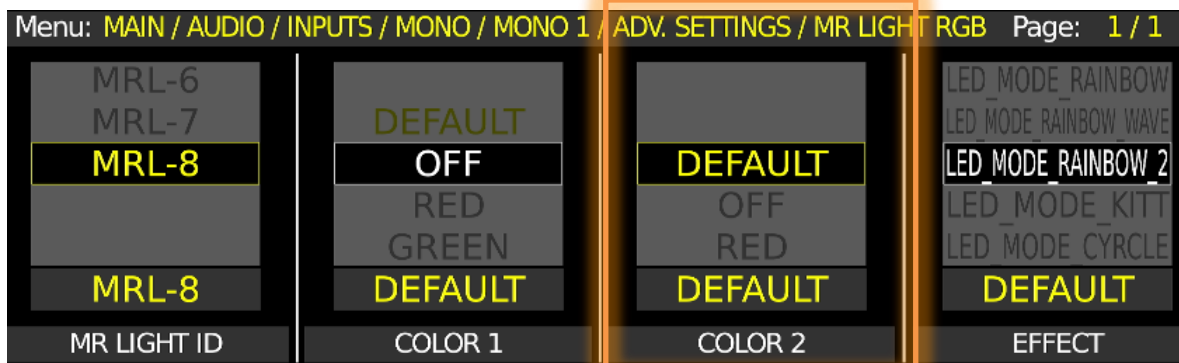
COLOR 1 selects the primary colour used by the MR Light RGB effect assigned to MONO.

The available selections are:

- **OFF:** **OFF** represents an unlit colour state. When **COLOR 1** is set to **OFF**: The primary phase of the effect produces no illumination / A **FIXED** effect may leave the lamp unlit
- **RED**
- **GREEN**
- **BLUE**
- **YELLOW**
- **CYAN**
- **MAGENTA**
- **WHITE**

The selected colour is produced by the RGB LEDs installed inside the lamp. Its perceived appearance can be affected by ambient lighting, the printed lamp panel and the configured LED brightness.

COLOR 2:

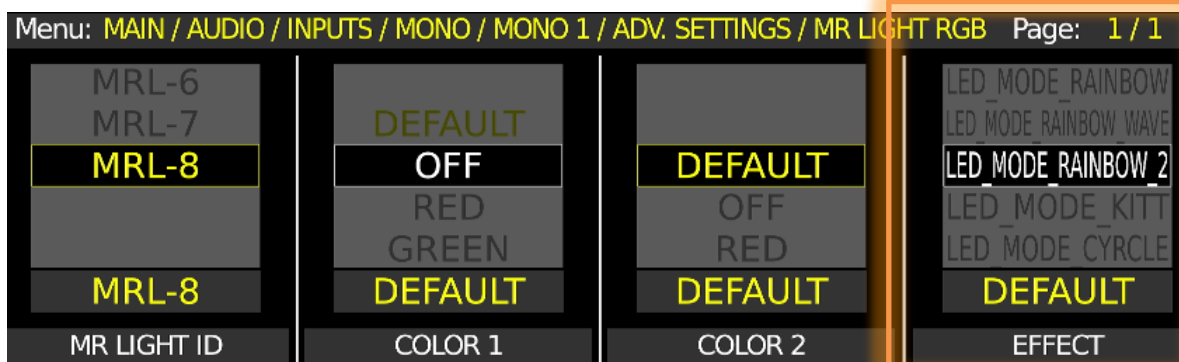


COLOR 2 selects the secondary colour used by lighting effects that require two colour states.

The selectable values are:

- OFF
- RED
- GREEN
- BLUE
- YELLOW
- CYAN
- MAGENTA
- WHITE

EFFECT:



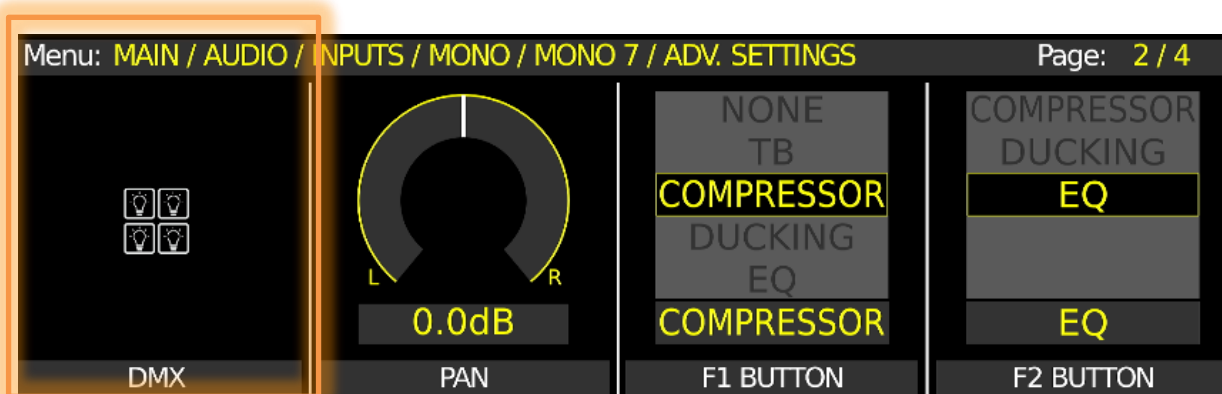
EFFECT determines how COLOR 1 and, where applicable, COLOR 2 are reproduced by the selected MR Light RGB device. The selectable EFFECT options are:

- **LED MODE OFF:** disables the LED output for the selected effect. In this mode: The lamp remains unlit / COLOR 1 and COLOR 2 do not produce a visible effect / No animation or transition is applied. This option is useful when the MR LIGHT assignment must remain stored but the lamp must not illuminate.
- **LED MODE FIXED:** produces a continuous, steady indication. In this mode: COLOR 1 remains continuously illuminated / No flashing or movement is applied

- **LED_MODE_BLINK:** produces a periodic flashing indication. Depending on the colour configuration: COLOR 1 may alternate with an unlit phase / COLOR 1 and COLOR 2 may alternate when two-colour operation is supported
- **LED_MODE_SINGLE_WAVE:** produces a single moving or sweeping light pattern across the illuminated area. The primary visible colour is determined by COLOR 1. COLOR 2 may be used as a secondary or background state according to the lamp firmware.
- **LED_MODE_DOUBLE_WAVE:** produces two simultaneous or sequential wave movements.
- **LED_MODE_FADING_WAVE:** **LED_MODE_FADING_WAVE** combines a moving wave with gradual brightness transitions. The illuminated area may progressively increase and decrease in intensity as it moves across the lamp. COLOR 1 is normally the principal colour, while COLOR 2 may define a secondary or background state.
- **LED_MODE_FADING_BLINK:** produces a flashing effect with gradual fade-in and fade-out transitions instead of abrupt switching.
- **LED_MODE_CIRCLE_WAVE:** produces a circular or ring-shaped wave movement. The light may appear to travel around the illuminated area in a circular sequence. COLOR 1 normally defines the principal wave colour, while COLOR 2 may be used as a secondary phase or background.
- **LED_MODE_NOISE:** produces an irregular or pseudo-random animated lighting pattern. This effect is visually more dynamic than fixed or regular flashing and should normally be reserved for special signalling or demonstration purposes.
- **LED_MODE_RAINBOW:** produces a continuously changing multicolour RGB sequence. The identifier uses the firmware spelling **RAIMBOW**.
- **LED_MODE_RAINBOW_WAVE:** combines a multicolour spectrum with a wave or sweeping movement. The lamp may display changing colours while the illuminated pattern travels across the device.
- **LED_MODE_RAINBOW_2:** is a second rainbow-style multicolour mode. It may use a different colour progression, timing, direction or spatial distribution from **LED_MODE_RAINBOW**.
- **LED_MODE_KITT:** produces a scanning effect similar to a moving light that travels across the lamp and returns in the opposite direction.
- **LED_MODE_CYRCLE:** produces a circular or cyclical lighting pattern.

The following descriptions explain the intended operating behaviour suggested by the firmware identifiers. The exact speed, direction, timing and colour handling are determined by the installed MR Light RGB firmware.

DMX:



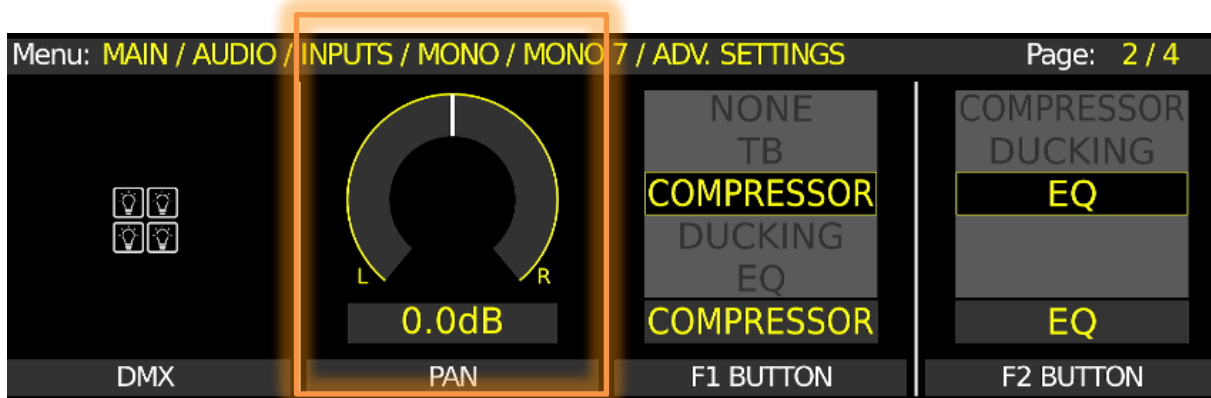
Associates the selected MONO source with the console's **DMX lighting-control system**. When enabled, opening or closing the source can generate a lighting event for compatible studio lights.

The available selections generally include:

- **OFF:** the source does not generate a DMX lighting event.
- **ENABLED/ASSIGNED EVENT:** associates the source status with a configured DMX channel, scene or lighting event.

The DMX address, colour, brightness and scene behaviour are configured in the corresponding lighting-control section.

PAN:

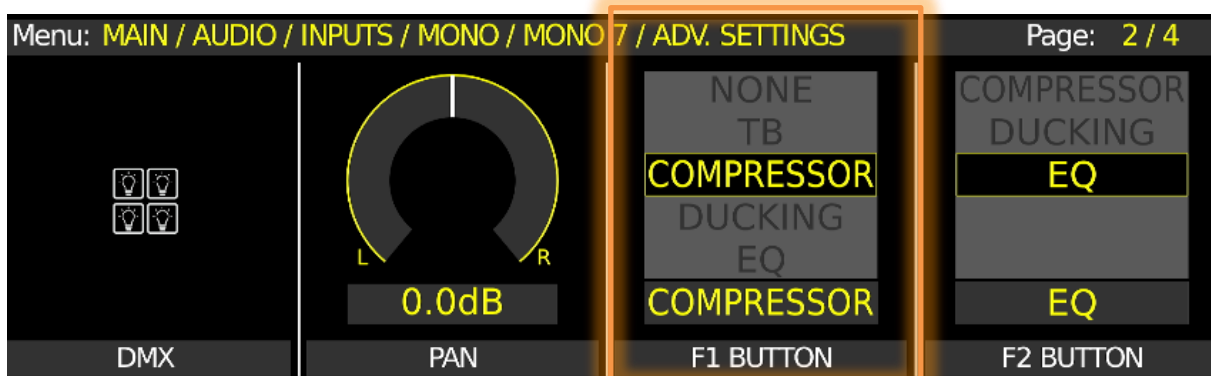


Positions the MONO signal within the stereo sound field of the output buses.

- **LEFT:** routes the signal mainly or completely to the left channel.
- **CENTER:** sends the signal equally to the left and right channels.
- **RIGHT:** routes the signal mainly or completely to the right channel.

Intermediate positions provide gradual level distribution between the two channels. PAN affects stereo buses such as PGM, SUB and stereo AUX outputs but does not convert the original MONO signal into a true stereo source.

F1 BUTTON:



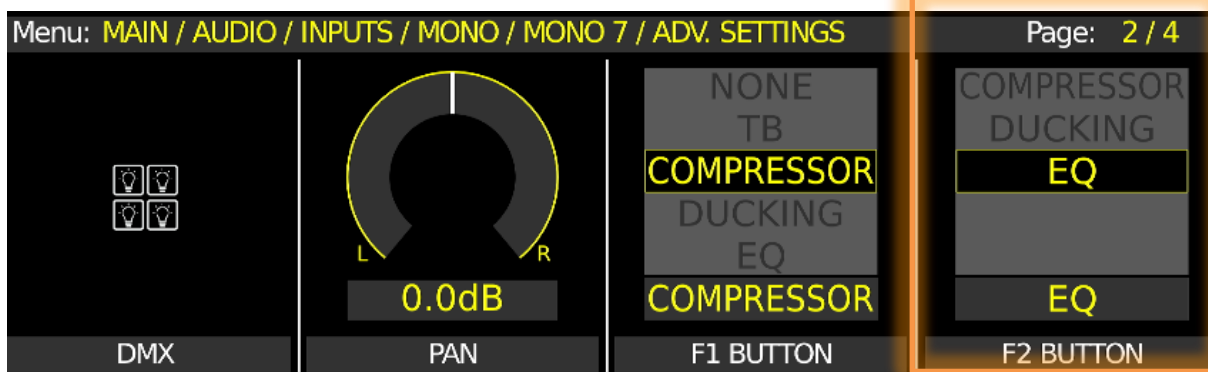
Defines the function assigned to the channel's **F1 button** when the selected MONO source is loaded.

Available options may include:

- **NONE:** no function is assigned.
- **TB:** activates Talkback communication.
- **EQ:** enables or disables the Equalizer when its Mode is UNLOCKED.
- **COMPRESSOR:** enables or disables Compressor processing.
- **DUCKING:** enables or disables the Ducking Master function when its Status is UNLOCKED.

The button LED provides visual feedback about the selected function's operating status.

F2 BUTTON:



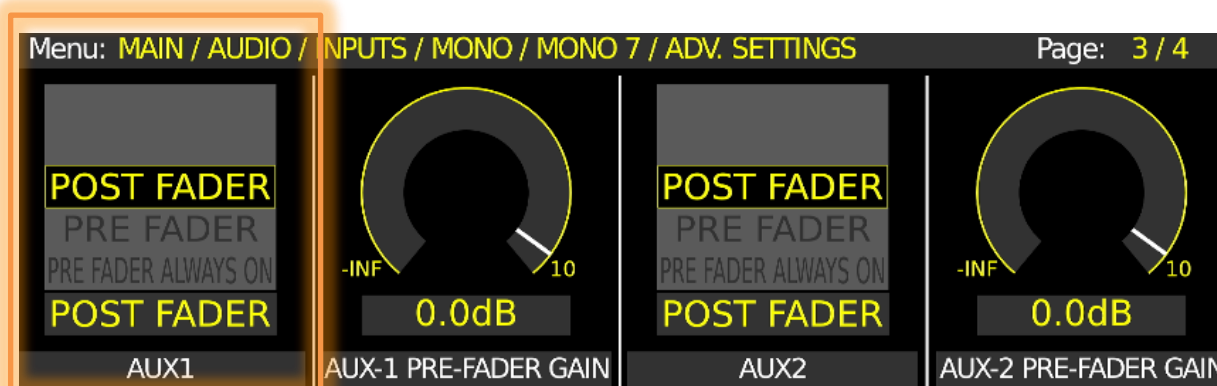
Defines the function assigned to the channel's **F2 button**. It provides the same selectable processing and communication functions available for F1, allowing two different commands to be controlled directly from the channel surface.

Available options may include:

- **NONE:** no function is assigned.
- **TB:** activates Talkback communication.
- **EQ:** enables or disables the Equalizer when its Mode is UNLOCKED.
- **COMPRESSOR:** enables or disables Compressor processing.
- **DUCKING:** enables or disables the Ducking Master function when its Status is UNLOCKED.

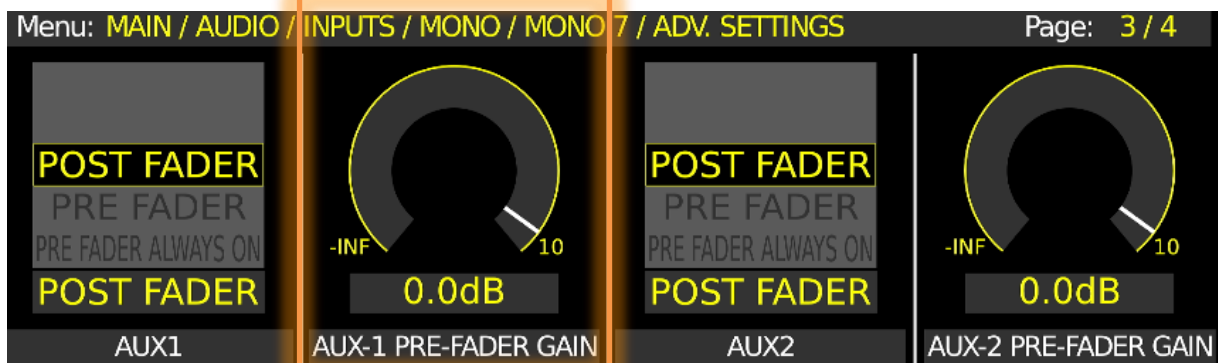
F1 and F2 can be configured independently.

AUX-1:

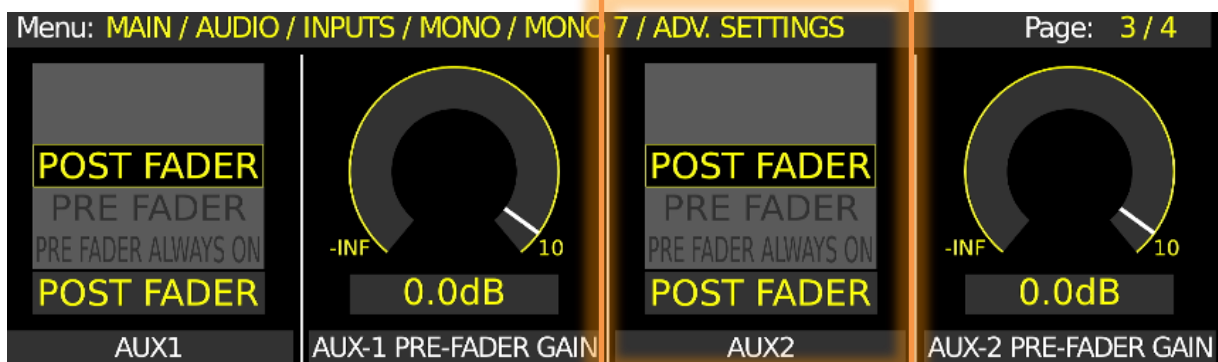


Defines how the selected MONO signal is routed to the **AUX-1 bus**.

- **POST FADER:** the AUX-1 signal follows the channel fader and ON/OFF status.
- **PRE FADER:** the AUX-1 signal is independent of the fader but follows the channel ON/OFF status.
- **PRE-FADER ALWAYS ON:** the AUX-1 signal remains active independently of both the fader position and ON/OFF status.

AUX-1 PRE-FADER GAIN:

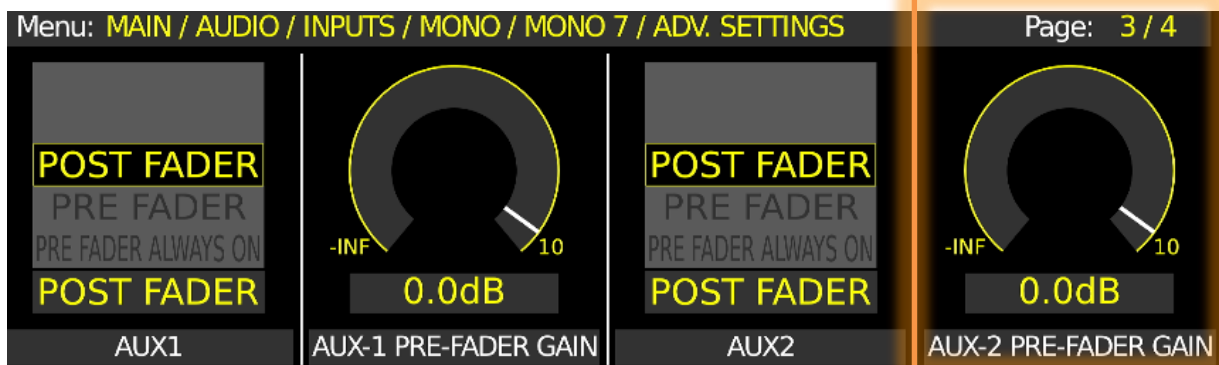
Adjusts the level sent to AUX-1 when a **Pre-Fader mode** is selected. It allows the auxiliary feed to be controlled independently from the main channel fader. This is useful for monitoring, recording, communication or external processing feeds. Excessive gain may cause **overload or clipping on AUX-1**.

AUX-2:

Defines how the selected MONO signal is routed to the **AUX-2 bus**.

- **POST FADER:** the AUX-2 signal follows the channel fader and ON/OFF status.
- **PRE FADER:** the AUX-2 signal is independent of the fader but follows the channel ON/OFF status.
- **PRE-FADER ALWAYS ON:** the AUX-2 signal remains active independently of the fader and ON/OFF status.

AUX-2 PRE-FADER GAIN:



Adjusts the independent level sent to AUX-2 when a **Pre-Fader mode** is selected. This setting does not change the signal level routed to PGM, SUB or other buses.

BUTTON LIGHT:



Defines the **RGB colour assigned to the channel buttons** when the selected MONO source is loaded. The illumination helps identify the source type and provides visual feedback about its operating status. The colour can be customized through the available **Red, Green and Blue values** or by selecting a configured colour preset, depending on the firmware interface. This setting affects only the console's visual indication and does not modify the audio signal.

FADER BAR LIGHT:

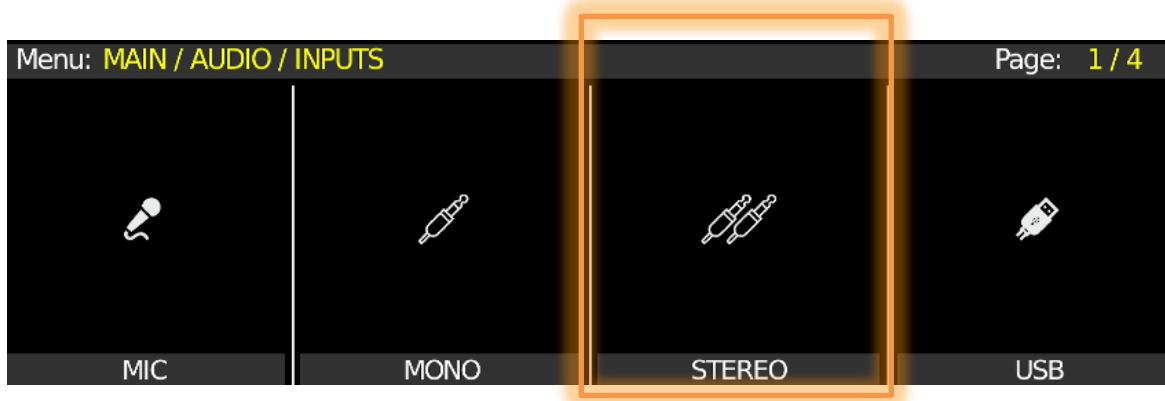
Defines the **RGB colour of the LED bar beside the channel fader**. The bar indicates the fader position and attenuation level, while its colour identifies the selected MONO source.

Using consistent colours for similar source types allows faster recognition during live operation. This parameter affects only the console surface illumination and **does not affect audio processing or routing**.

STEREO

A STEREO input carries two related audio channels: **Left (L)** and **Right (R)**. These channels are processed together to preserve the spatial information and stereo image of the original source.

On the Oxygen 3000 Plus, stereo inputs are intended for professional line-level devices such as audio players, radio-automation systems, computers, codecs and external receivers. The analogue stereo inputs are balanced, with a nominal level of **0 dBu**, a maximum input level of **+18 dBu** and 24-bit/48 kHz conversion.

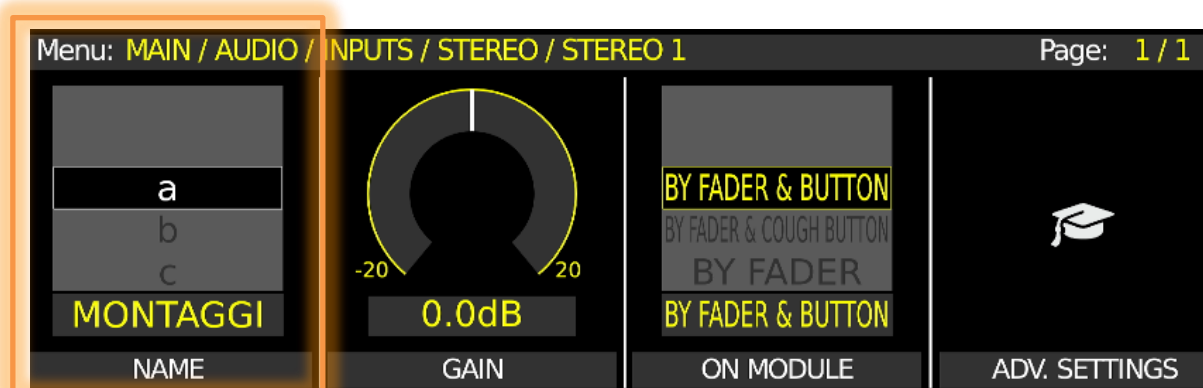


ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

INPUTS SETTING (STEREO)

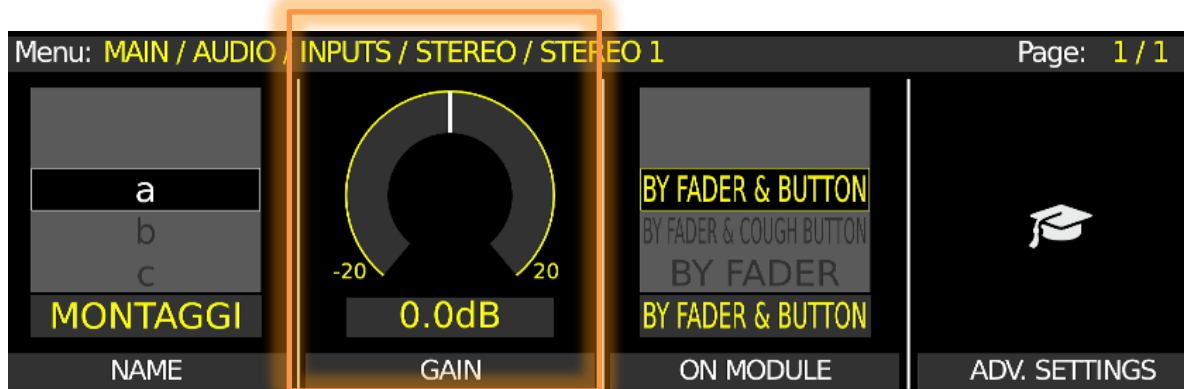
In this page you can set different options: **NAME, GAIN, ON MODULE, ADVANCED SETTINGS.**

NAME:

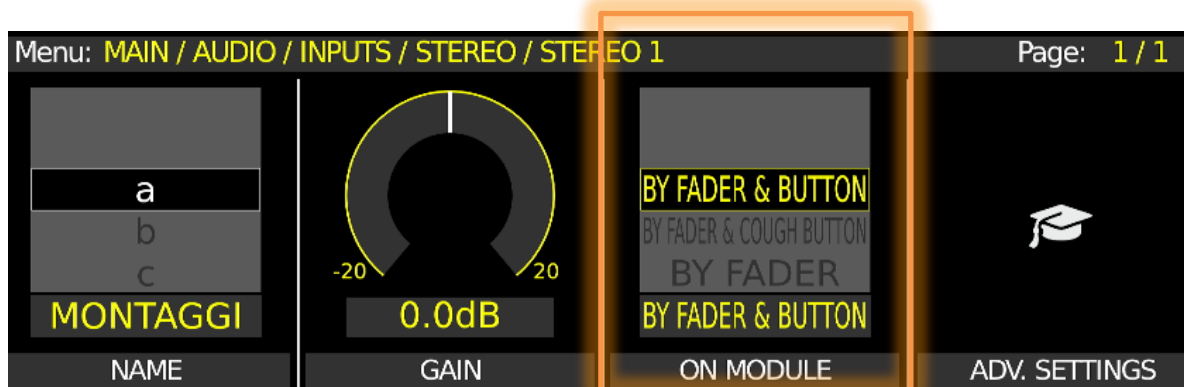


Assigns a customized identification label to the selected STEREO source. The label is displayed in the source-selection menus and on the channel display when the source is assigned to a mixer channel.

The name can contain up to **8 characters**. Press **ESC** to complete the editing procedure.

GAIN:

Adjusts the digital level of the selected STEREO source. The same gain correction is applied to both Left and Right channels, preserving their relative balance and stereo image. Positive values increase the input level, while negative values attenuate it. Excessive gain may cause digital clipping; insufficient gain may produce a weak output signal. **Adjustment range: -20 dB to +20 dB. Step: 0.1 dB.**

ON MODULE:

Defines how the selected STEREO source is switched on or off when assigned to a mixer channel or associated control module.

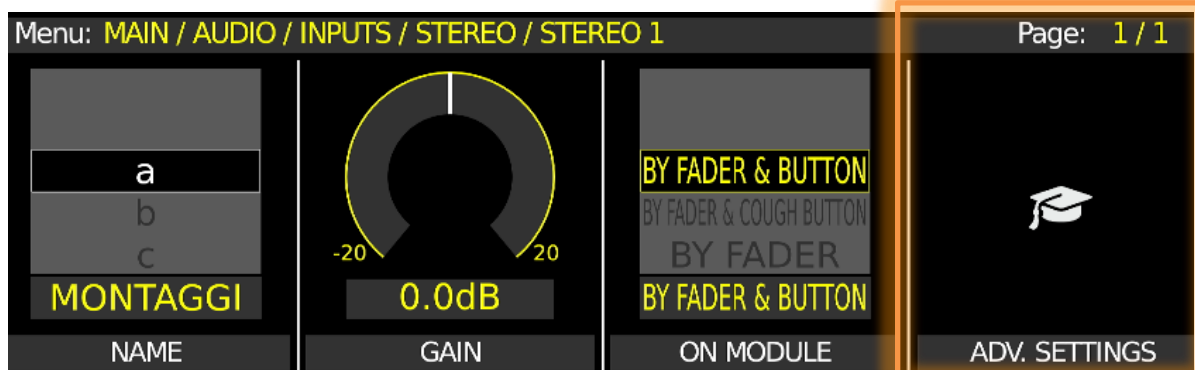
- **FADER & BUTTON:** the source status can be controlled through both the channel fader and the ON/START button.
- **BY FADER:** the source is switched on when the fader crosses the activation threshold and switched off when it moves below that threshold.
- **BY BUTTON:** the source status is controlled by the ON/START button. Moving the fader does not automatically change its ON/OFF status.
- **BY FADER & COUGH BUTTON:** the fader controls normal activation, while the Cough button temporarily mutes the source. Releasing the button restores the previous operating condition.
- **ALWAYS ON:** the STEREO source remains permanently enabled.
- **ALWAYS OFF:** the STEREO source remains permanently disabled.

The **ALWAYS ON** option is useful for permanently available feeds but should be used carefully because the source remains active independently of normal surface controls.

ADV. SETTINGS:

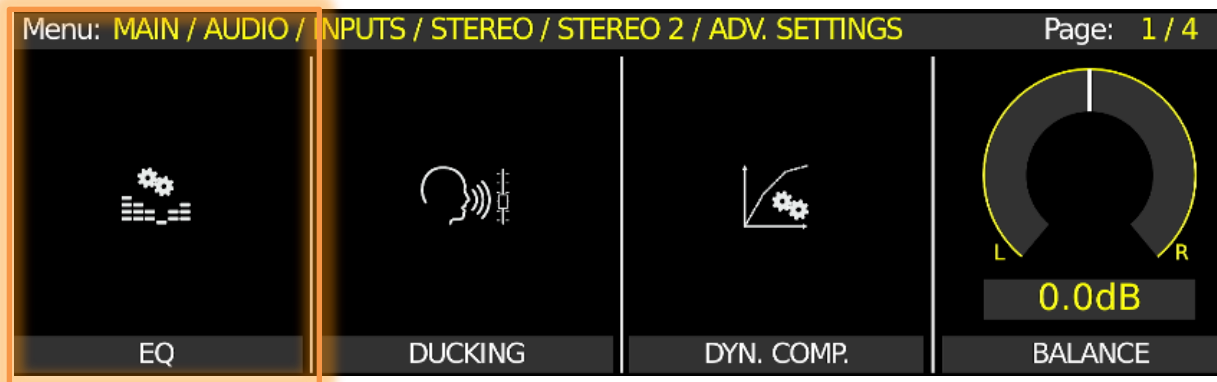
Opens the advanced configuration section for the selected STEREO source. It provides access to additional **audio processing, routing, external-control and surface-illumination parameters**.

Unlike MIC and MONO inputs, STEREO sources do not normally include the dedicated Noise Gate section, because the Oxygen 3000 Plus dynamic processor is intended primarily for **microphone and mono inputs**.



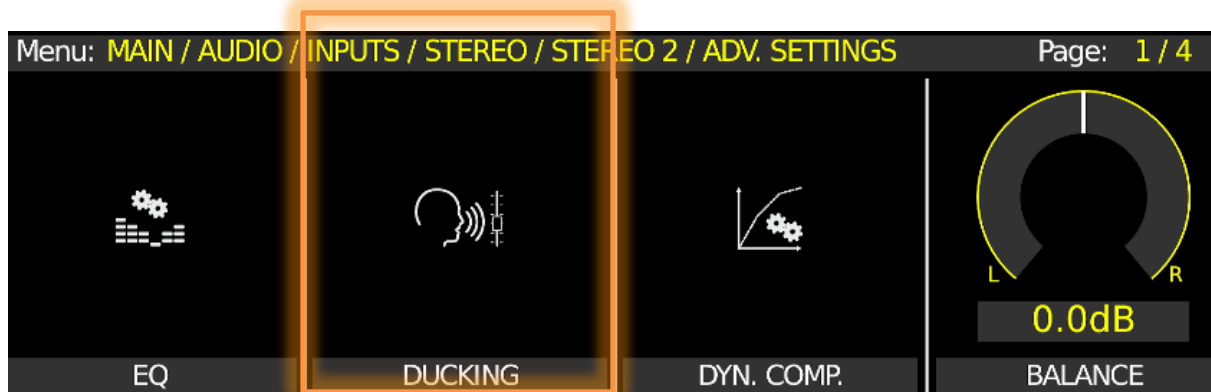
In ADV. SETTING you can find a lot of different parameters: **EQ, DYN. COMP, DUCKING, BALANCE, MODE, F1 MODE F2 MODE, AUX 1, AUX 1 PRE-FADER GAIN, AUX 2, AUX 2 PRE-FADER GAIN, BUTTON LIGHT, FADER BAR LIGHT.**

EQ:

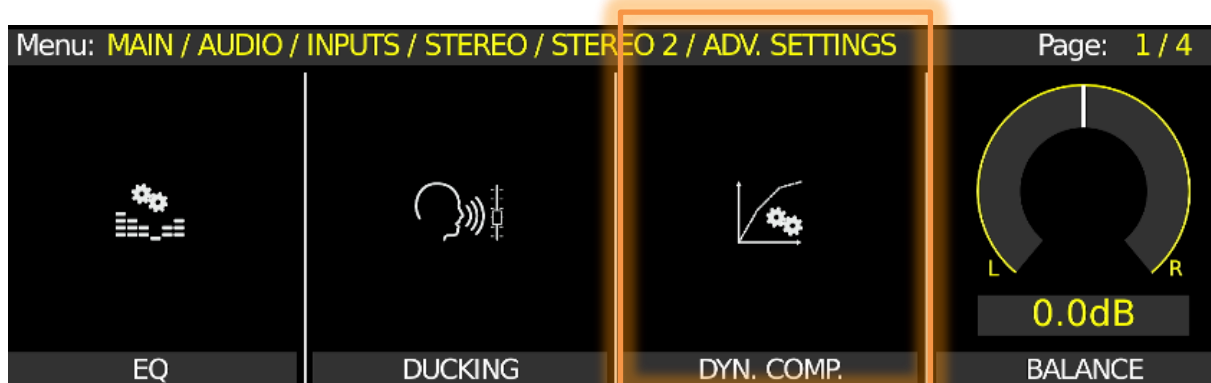


The **EQ** section provides a multi-band parametric equalizer for adjusting the tonal balance of the stereo source. The same equalization curve is applied to the stereo channel, preserving the relationship between the left and right signals. The Oxygen equalizer covers a frequency range from approximately **31.5 Hz to 20 kHz**.

ATTENTION: A detailed description of this mode can be found in the section dedicated to the MIC input

DUCKING:

The **Ducking** function automatically attenuates the stereo input when another source—normally a microphone—is active. A typical application is the automatic reduction of background music while the presenter is speaking. **SLAVE MODE:** determines whether the selected stereo input is attenuated by the ducking system. **Slave Mode enables or disables ducking on the stereo source.**

DYN. COMP.:

The **DYN. COMP.** section controls the dynamic processing applied to the stereo input. It can be used to reduce excessive level variations, prevent signal overloads and attenuate unwanted low-level noise.

The section normally includes:

- **DYNAMIC PRESET:** recalls or stores a predefined set of dynamic-processing parameters.
- **COMPRESSOR:** reduces the level of signals exceeding the configured threshold. The main parameters include **Threshold, Ratio, Attack, Release, Hold, Limiting Threshold and Makeup Gain.**

Where available, the processor is applied to the stereo source in a coordinated manner, helping to prevent unwanted changes in the stereo image.

ATTENTION: A detailed description of this mode can be found in the section dedicated to the MIC input

BALANCE:



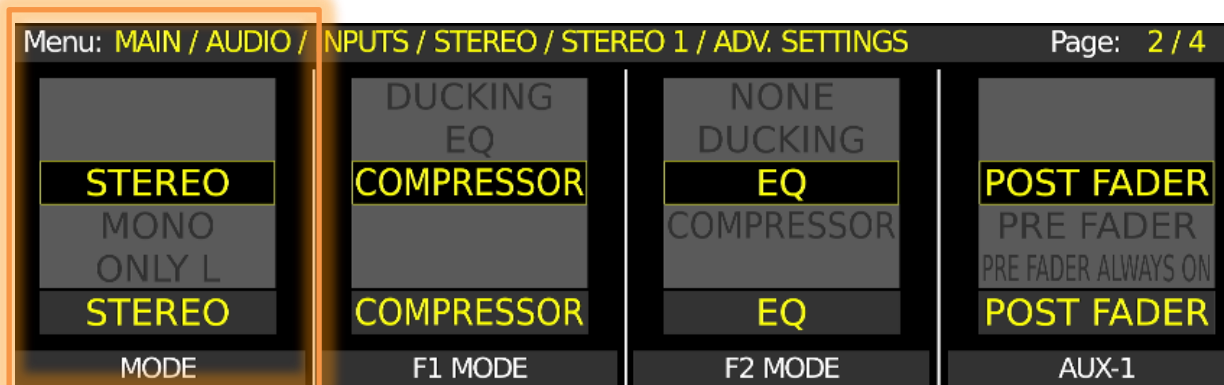
The **Balance** parameter adjusts the relative level of the left and right channels without changing the overall input assignment.

The available positions range from:

- **LEFT:** progressively attenuates the right channel.
- **CENTER:** applies equal level to the left and right channels.
- **RIGHT:** progressively attenuates the left channel.

The parameter is adjusted in **1.0 dB steps**, allowing accurate correction of level differences between the two sides of the stereo signal.

MODE:



The **Mode** parameter determines how the left and right components of the input are processed and routed. The selectable options are:

- **STEREO:** preserves the original left and right channels.
- **MONO:** combines the left and right signals into a mono signal.
- **L:** uses only the left input channel.
- **R:** uses only the right input channel.
- **INV-L:** reverses the polarity of the left channel while leaving the right channel unchanged.
- **INV-R:** reverses the polarity of the right channel while leaving the left channel unchanged.
- **INV. L&R:** reverses the polarity of both left and right channels.

The polarity-inversion modes can help correct **phase or wiring problems** in connected audio sources.

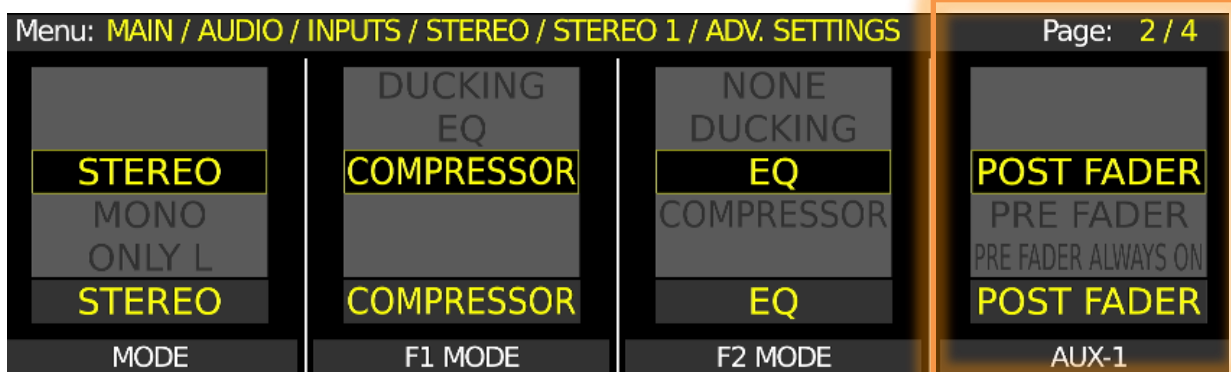
F1 MODE / F2 MODE:

The **F1 Mode** and **F2 Mode** parameters assign a specific function to the two programmable buttons on the channel strip.

Depending on the installed firmware and the functions enabled for the source, the selectable options may include:

- **NONE:** no function is assigned.
- **EQ:** enables or disables the equalizer.
- **COMPRESSOR:** enables or disables the compressor.
- **DUCKING:** enables or disables the permitted ducking function.

If a processor is set to **LOCKED ON** or **LOCKED OFF**, its assigned F button cannot override that status.

AUX 1 / AUX 2:

The **AUX 1** and **AUX 2** parameters determine where the stereo input signal is taken in relation to the channel fader and ON/START control before being sent to the corresponding auxiliary bus.

The selectable modes are:

- **POST-FADER:** the AUX level follows the channel fader. Closing the fader reduces or removes the signal from the AUX bus.
- **PRE-FADER:** the AUX signal is independent of the channel fader but remains subject to the channel's ON/START status.
- **PRE-FADER ALWAYS ON:** the AUX signal is independent of both the channel fader and the ON/START button. The source remains present on the AUX bus even when the channel is closed.

POST-FADER is normally used when the AUX level must follow the main mix, while **PRE-FADER** is suitable for independent monitoring, recording or service feeds.

AUX 1 PRE-FADER GAIN / AUX 2 PRE-FADER GAIN:

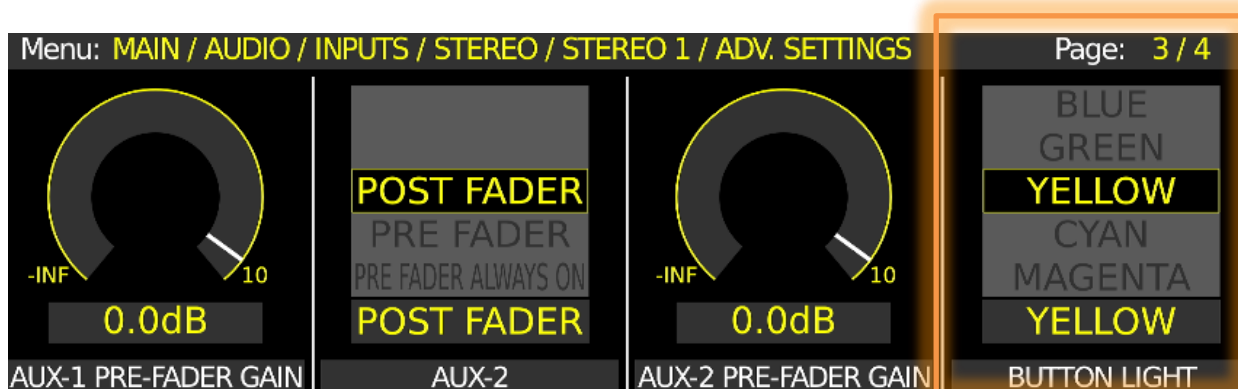


These parameters adjust the level sent to **AUX 1** or **AUX 2** when the corresponding auxiliary bus is configured in a pre-fader operating mode.

- A **negative value** attenuates the signal sent to the AUX bus.
- **0 dB** applies unity gain.
- A **positive value** increases the AUX send level.

This adjustment allows the operator to create an **independent auxiliary mix without changing the main channel-fader level**.

BUTTON LIGHT:



The **Button Light** parameter selects the RGB colour used for the channel's **ON/START button**.

The selected colour helps the operator:

- identify the source category;
- distinguish the channel from adjacent inputs;
- recognise its operating status quickly.

The button can display different brightness or colour states to indicate whether the channel is **off, ready or active**.

FADER BAR LIGHT:



The **Fader Bar Light** parameter selects the RGB colour of the LED bar located next to the channel fader.

The LED bar provides a visual indication of:

- the **fader position or attenuation level**;
- the **source type or channel identification**;
- the selected channel colour.

The available RGB colours can be configured according to the station's operating conventions, providing **clear and immediate visual identification of the stereo input**.

USB AUDIO

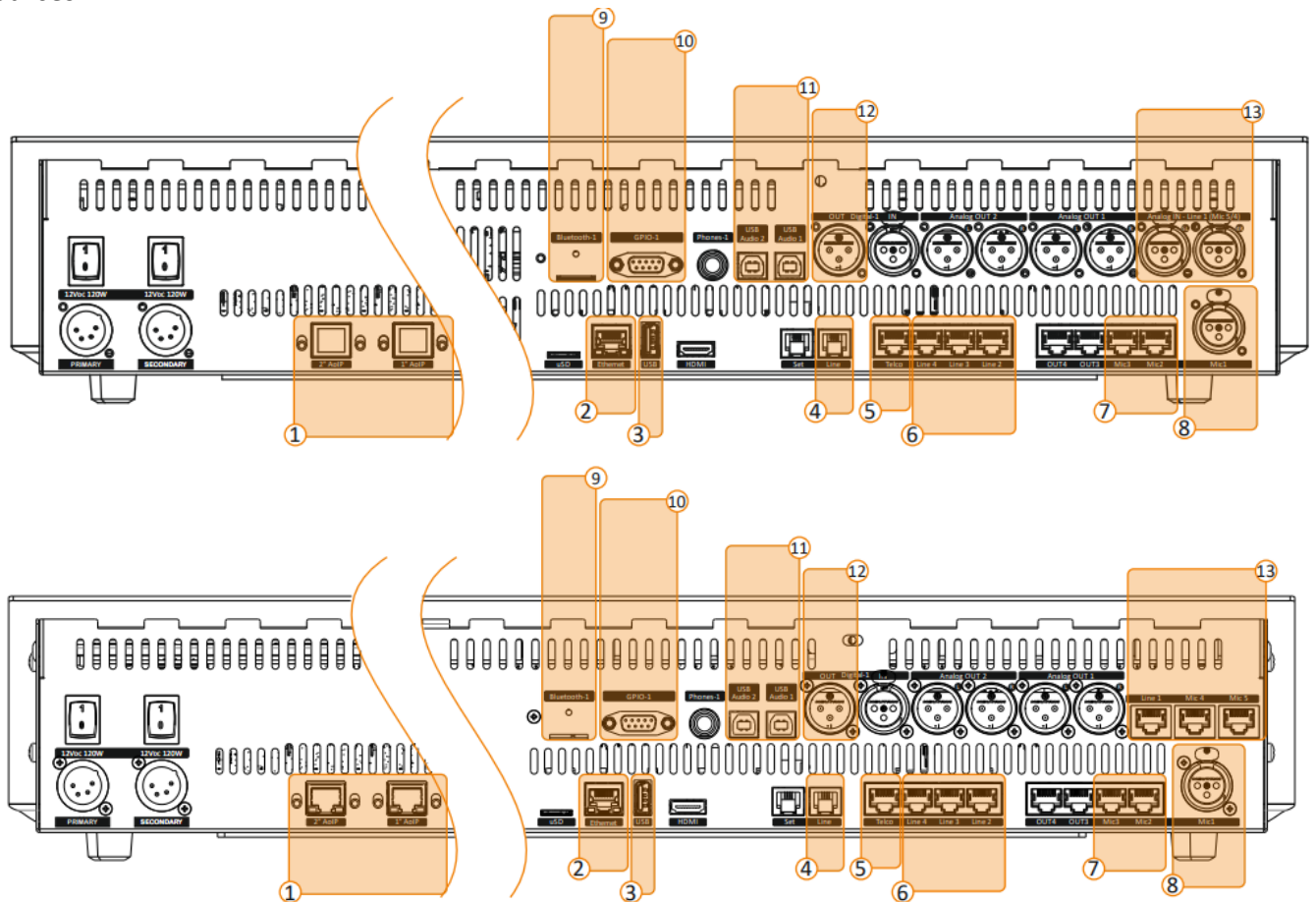
2 USB audio interface per each board

The USB audio interface allows you to directly connect the PC to the **Oxygen** console, with no need for audio cards. In fact, the PC detects the console as a digital audio card with 2 stereo inputs and 1 stereo outputs for simultaneous playout and recording (**Bidirectional**).

USB-1/3 Input 24bit cards

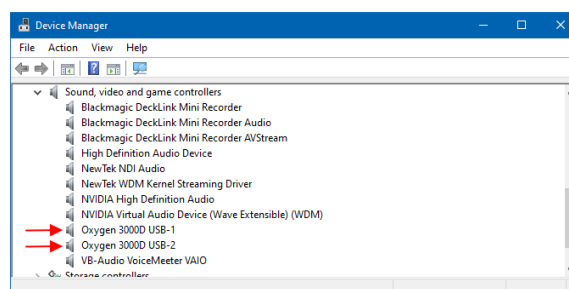
USB-2/4 Input & Output 16bit cards

two Built-in stereos **USB I/O** Audio Interface to connect directly to a computer. USB Audio Card with a connector **Type-B**. With this type of connection, you can save hundreds of dollars on an audio card. By **OXYGEN** Digital Mixing Console, you can connect your computer or any digital device via perfect USB audio **I/O** sources.



Important: The audio input connections for **USB Audio** are located on the first Audio Board, at position **11**.

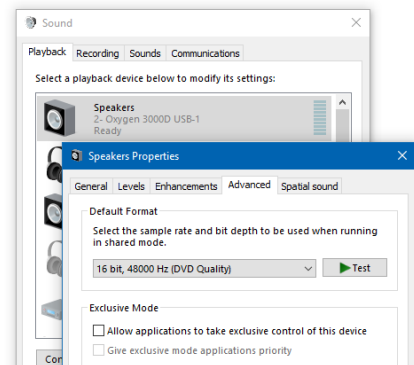
Once the mixer is connected to a computer, it will be recognized automatically and will not require any intervention. The connection is made via a normal **USB-B cable 2.0**



NOTE: Highly recommend not to use a (SS 10) SUPERSPEED USB port on the PC side.



To set the parameters correctly, set the **input/recording** and **output/playback** as default (**48000 Hz**).



ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

INPUTS SETTING (USB AUDIO)

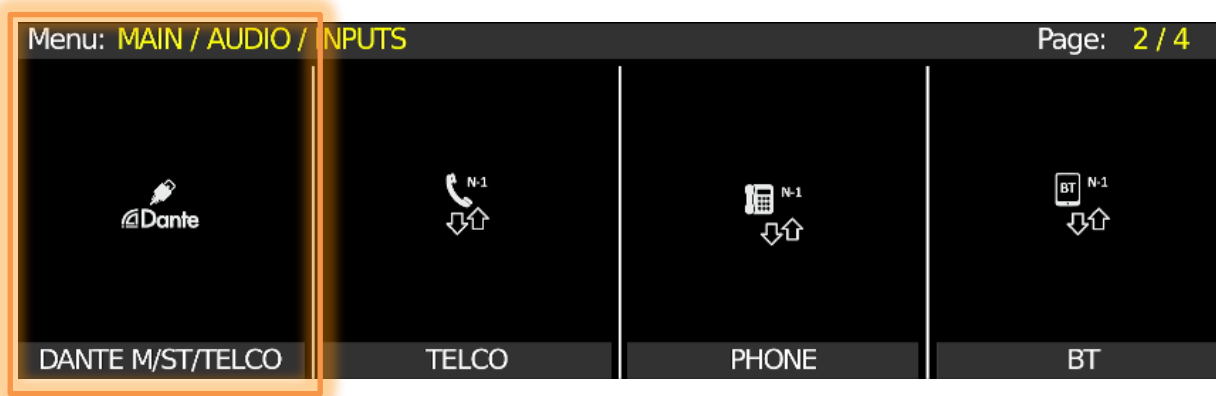
Here we can find the source parameters (USB AUDIO). In this page you can modify/set different parameters of the USB AUDIO. The main modality are: **NAME, GAIN, ON MODULE** and **ADV SETTINGS**.

MAIN / AUDIO / INPUTS / USB / USB 2

ATTENTION: The main settings available in this interface, including those found under **ADV. SETTINGS**, are identical to the settings provided for the **STEREO input**.

DANTE* (OPTIONAL)

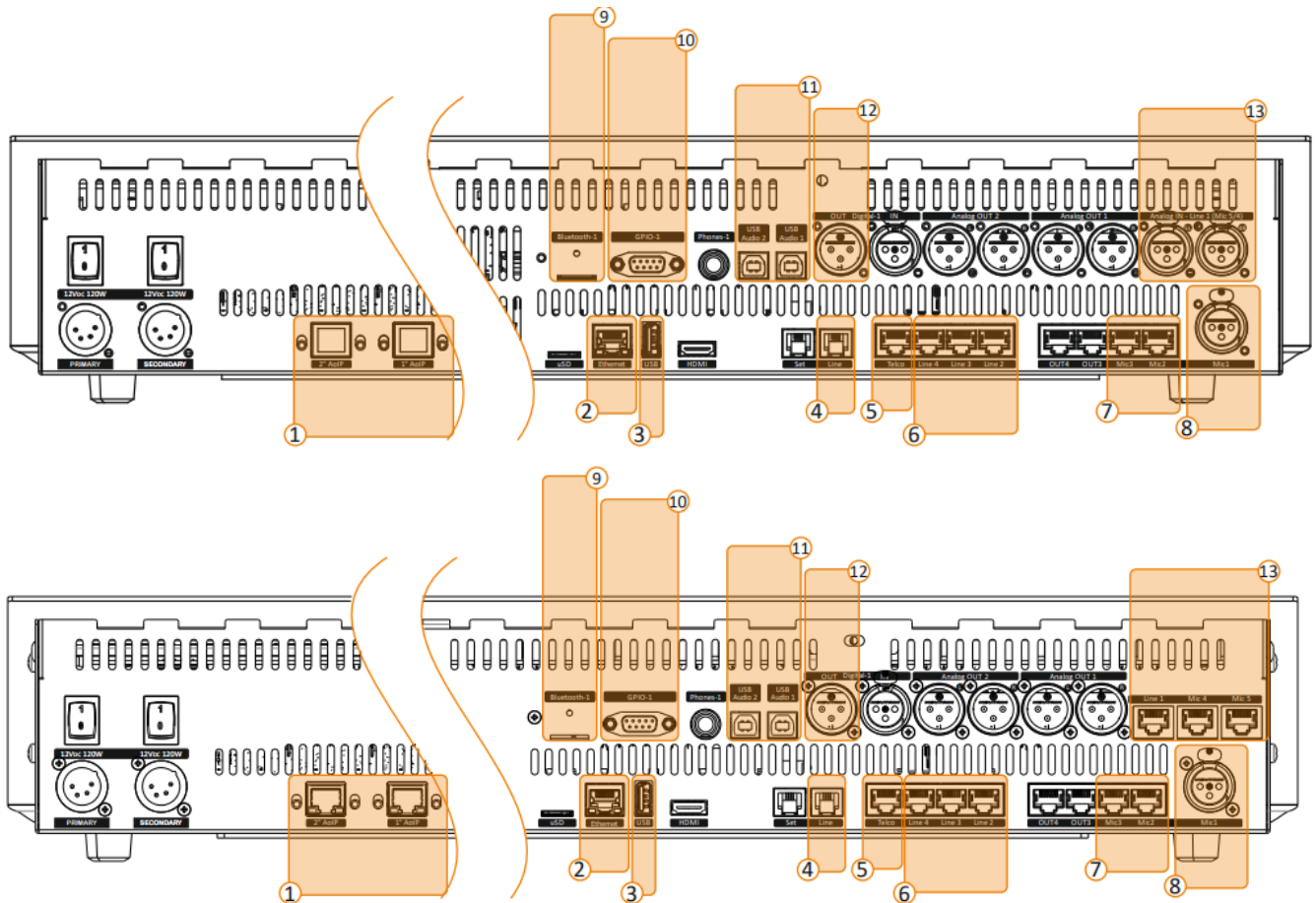
The **Dante input source** allows the Oxygen 3000 Plus to receive professional digital audio signals over a standard Ethernet network. Instead of using separate analogue or AES/EBU cables for each signal, multiple audio channels can be transported through a single network connection with **low latency, precise clock synchronisation and digital audio quality**. The optional Dante interface is **AES67 compliant** and provides an Ethernet connection for **16 or 32 mono inputs** (8 or 16 stereo inputs) and **16 or 32 mono outputs** (8 or 16 stereo outputs), depending on the installed version. Each signal has **independent and dedicated level control**, while **Sample Rate Conversion (SRC)** helps integrate sources operating with the supported sample rates. The interface supports **44.1, 48, 88.2 and 96 kHz** sample rates and **16-, 24- or 32-bit** audio. The latest Oxygen 3000 Plus specifications also indicate support for **SMPTE ST 2110-30** on compatible and appropriately configured devices. The **Primary LAN port** is the main connection to the Dante audio network. The **Secondary LAN port is not available**; consequently, this interface does **not provide Dante redundant-network operation**. Dante connectivity is optional and must normally be requested when ordering the console.



To connect a Dante source to the Oxygen 3000 Plus:

1. Connect the console's **Dante Primary LAN port** to the Dante network using a suitable Ethernet cable. A **Gigabit Ethernet switch** is recommended when several Dante devices are connected. A direct connection may be used for a simple two-device configuration.
2. Connect a computer running **Dante Controller** to the same network. Dante devices are normally discovered automatically.
3. In Dante Controller, open the **Routing** view. Locate the source device under **Transmitters** and the Oxygen 3000 Plus under **Receivers**.
4. Click the required routing cross-points to connect the transmitting channels to the Oxygen receiving channels. A **green check mark** confirms a successful subscription.
5. Check that the devices use compatible **sample-rate, clock and audio-format settings**. Incorrect settings may prevent the subscription from being established correctly.
6. On the console, select the required **DANTE source**, choose its input configuration and assign it to the desired mixer module.

Important: the Dante audio port is a **dedicated Audio-over-IP connection**. It should not be confused with the Ethernet connection used for general console control or configuration.

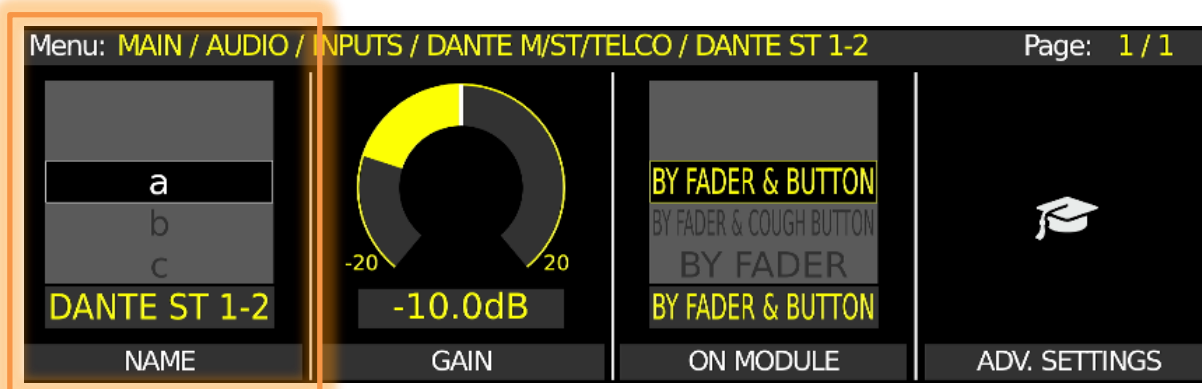


Important: The audio input connections for **Dante** are located on the first Audio Board, at position **1**. Dante card is available (as an option) with one Ethernet port or dual Ethernet ports.

INPUTS SETTING (DANTE)

Here we can find the source parameters (AESEBU). In this page you can modify/set different parameters of the **DANTE**. The main modality are: **NAME**, **GAIN**, **ON MODULE** and **ADV SETTINGS**.

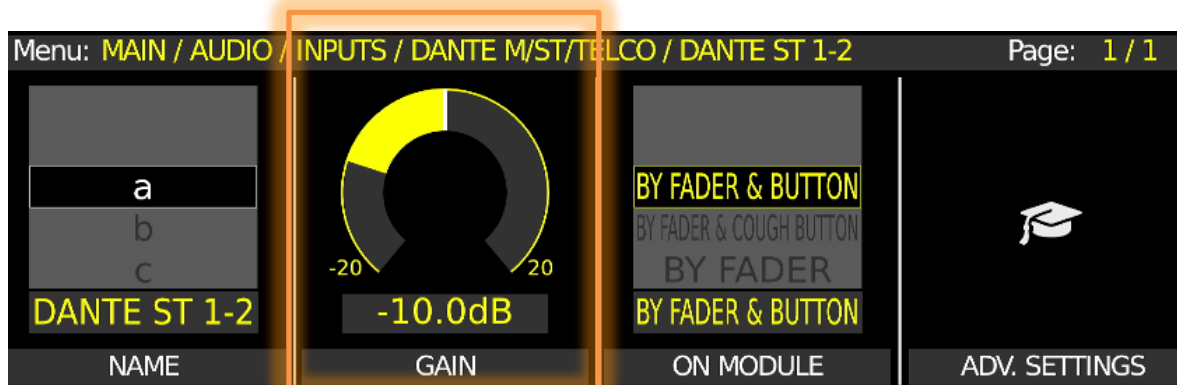
NAME:



NAME assigns a customised label to the selected Dante input. The name can contain up to **8 characters**. Press **ESC** to complete the editing procedure. A descriptive name such as **DANTE STUDIO**, **DANTE PLAYER**, **DANTE CODEC** or **DANTE TELCO** helps the operator identify the source quickly.

Important: changing NAME in the OXYGEN menu does not rename the Dante device or its Tx/Rx channels in Dante Controller. Dante network names must be managed separately through Dante Controller.

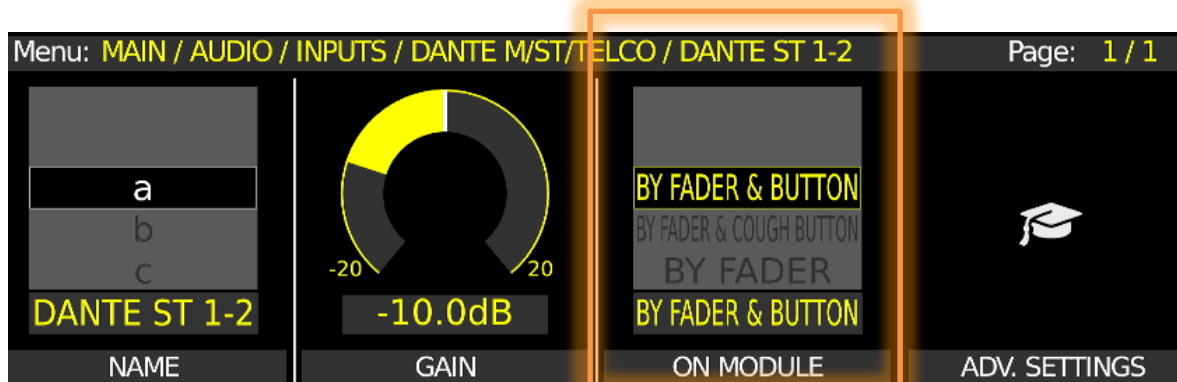
GAIN:



GAIN adjusts the digital input level of the selected Dante source before it enters the channel's processing and routing stages. **Range:** -20.0 dB to 20.0 dB. **Adjustment Step:** 0.1 dB. Reducing the gain can prevent an excessively high network source from overloading the console's internal processing. Increasing it can compensate for a source received at a low level. The GAIN value is associated with the selected Dante source rather than with the physical fader. Therefore, the stored value follows the source when it is assigned to another channel.

Important: increasing GAIN cannot recover information already lost because of clipping in the transmitting device. The level should first be adjusted correctly at the Dante transmitter.

ON MODULE:



ON MODULE determines how the selected Dante input is enabled or disabled on the console.

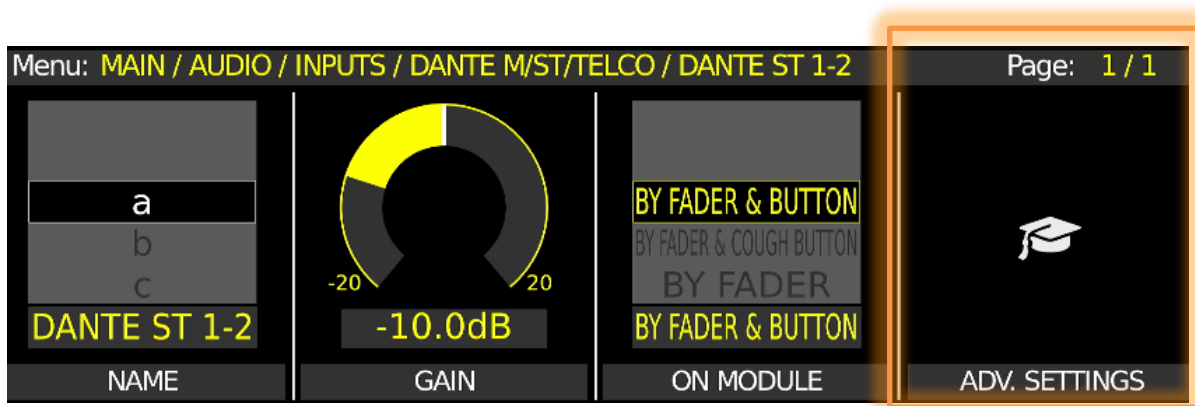
The selectable options are:

- **FADER & BUTTON:** The source is controlled through the combined operation of the channel **fader** and **ON/START button**. This mode provides manual control from the button while also allowing the fader position to participate in the channel's opening and closing logic.

- **BY FADER:** The source is activated when the fader passes the configured opening threshold and is deactivated when the fader returns below it. This mode is useful when the operator must place the source On Air by moving the fader only.
- **BY BUTTON:** The source is activated and deactivated exclusively through the channel's **ON/START button**. The fader controls the audio level but does not independently change the module's ON/OFF state.
- **BY FADER & COUGH BUTTON:** The source is activated by the fader position, similarly to **BY FADER**, but can be temporarily muted using the channel's **COUGH button**. This mode is typically used for microphone channels, allowing the operator to:
 - Open the microphone by raising the fader.
 - Temporarily mute the signal without changing the fader position.
 - Quickly restore the audio by releasing the COUGH button.
 The COUGH function provides a fast and silent muting control, useful during live speech or broadcast situations.
- **ALWAYS ON:** The Dante input module remains permanently enabled, independently of the fader and ON/START button. This mode is useful for continuously available sources, monitoring feeds or signals routed to auxiliary paths. The fader and bus assignments may still affect the level and destination of the signal.
- **ALWAYS OFF:** The Dante input module remains disabled and cannot be placed On Air through the normal fader or ON/START controls. This option can be used to prevent an unused or temporarily unavailable Dante source from being activated accidentally.

Important: ON MODULE controls the console input logic. It does not create, remove or mute the Dante subscription at network level.

ADVANCED SETTINGS:



ADV. SETTINGS opens the extended configuration page for the selected Dante input. The available parameters depend on whether the source is configured as **DANTE ST**, **DANTE MONO** or **DANTE TELCO**. They include:

- **EQ**
- **DYN. COMP**
- **DUCKING**
- **PAN or BALANCE**
- Input-channel **MODE**

- **F1 MODE and F2 MODE**
- **AUX 1 and AUX 2 routing**
- **AUX PRE-FADER GAIN**
- **BUTTON LIGHT**
- **FADER BAR LIGHT**

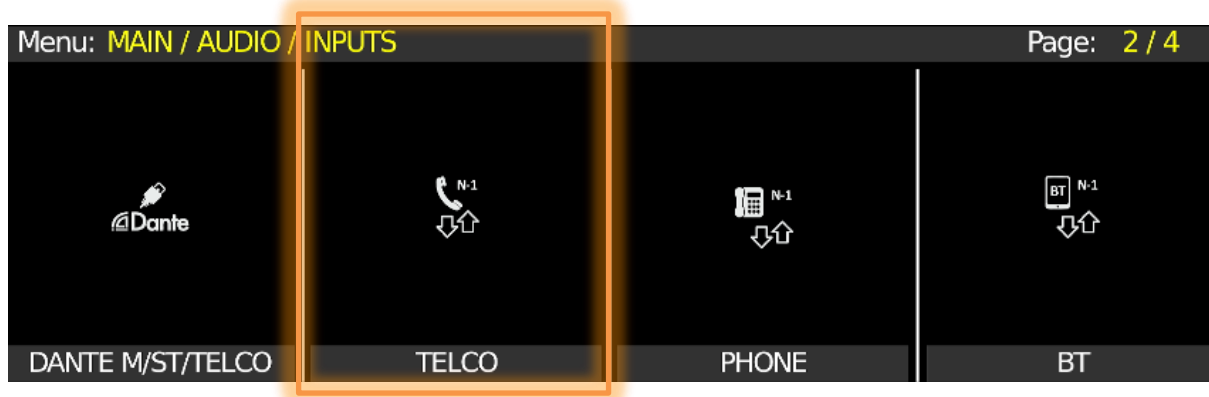
ATTENTION: For detailed information about the **ADVANCED SETTINGS**, refer to the corresponding **MONO**, **STEREO**, or **TELCO** input section, depending on the selected Dante configuration: **DANTE M**, **DANTE ST**, or **DANTE TELCO**, respectively.

TELCO

The **TELCO** section configures the input and return paths associated with an external telephone hybrid connected to the OXYGEN 3000 Plus. Depending on the installed firmware, the path may instead be displayed as: **MAIN → AUDIO → INPUTS → TELCO → TELCO 1**

Unlike **TELEPHONE**, which uses the console's internal POTS/PSTN hybrid, TELCO is designed to work with an external telephone hybrid, codec or compatible call-management device.

The dedicated TELCO I/O RJ45 connection can carry the two audio signals and the two control signals required by an external hybrid. Additional TELCO sources may also be created from analogue or Dante input pairs configured in **2 TELCO** mode.



The TELCO page includes:

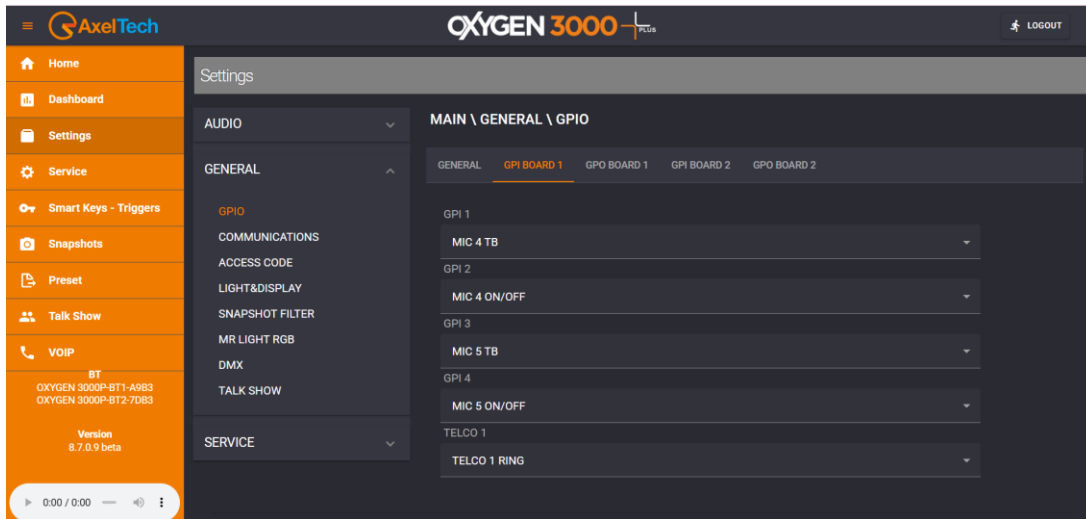
- **NAME**
- **GAIN RX**
- **GAIN TX**
- **ON MODULE**
- **ADV. SETTINGS**
- **DEVICE**

Important: TELCO does not contain the telephone-line interface itself. The external hybrid remains responsible for connecting to the POTS, GSM, VoIP or other telephone service.

ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

TELCO 1 GPIO SETTINGS:

The GPIO functions associated with the primary TELCO 1 device can be configured through the following OXYGEN REMOTER submenu: **MAIN / GENERAL / GPIO**



To configure the TELCO 1 GPI signal, assign the **RING** function through the following submenu:
MAIN / GENERAL / GPIO / GPI / TELCO 1

The GPI signal receives the **RING** status from the external TELCO device and forwards it to the console.

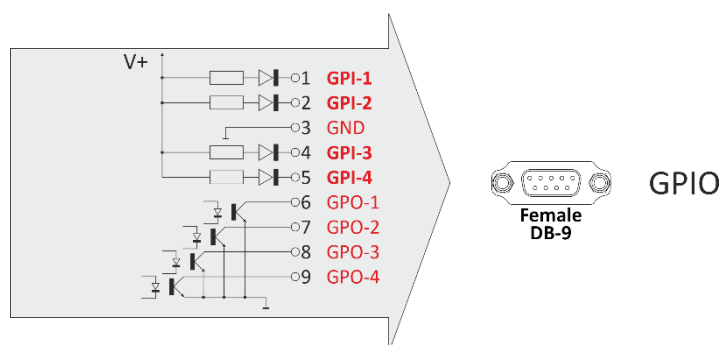
To configure the TELCO 1 GPO signal, assign the **HOOK** function through the following submenu:
MAIN / GENERAL / GPIO / GPO / TELCO 1

The GPO signal sends the **HOOK** command from the console to the external TELCO device.

GPIO CONNECTIONS FOR ADDITIONAL EXTERNAL TELCO DEVICES:

If additional external TELCO devices need to be connected, any GPIO signals that have not already been assigned to TELCO 1 or to other console functions can be used. The available GPI and GPO signals are provided through the SUB-D9 GPIO connector. The pin assignment of this connector is shown in the GPIO PINOUT diagram below. Each available GPI and GPO signal can be individually configured through the following menu: **MAIN / GENERAL SET. / GPIO**

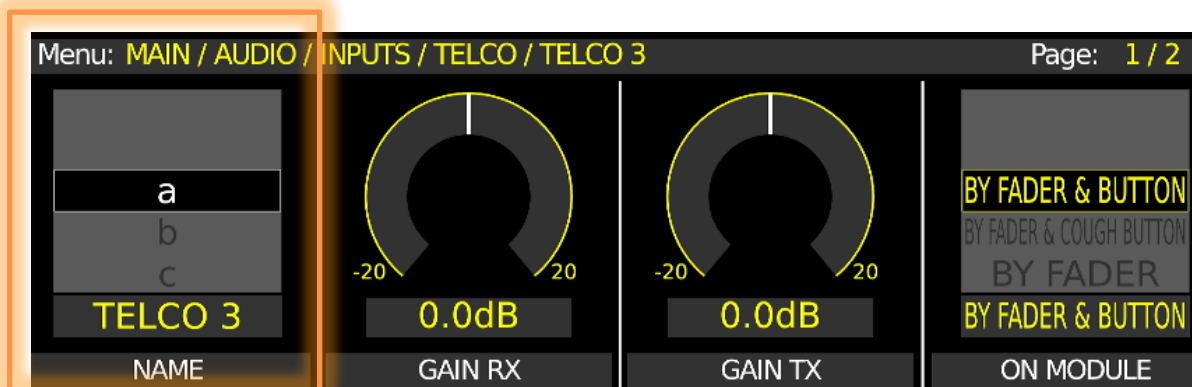
This configuration applies only to additional external TELCO devices. The RING and HOOK assignments for the primary TELCO 1 device are described in the previous section.



ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

INPUTS SETTING (TELCO)

NAME:



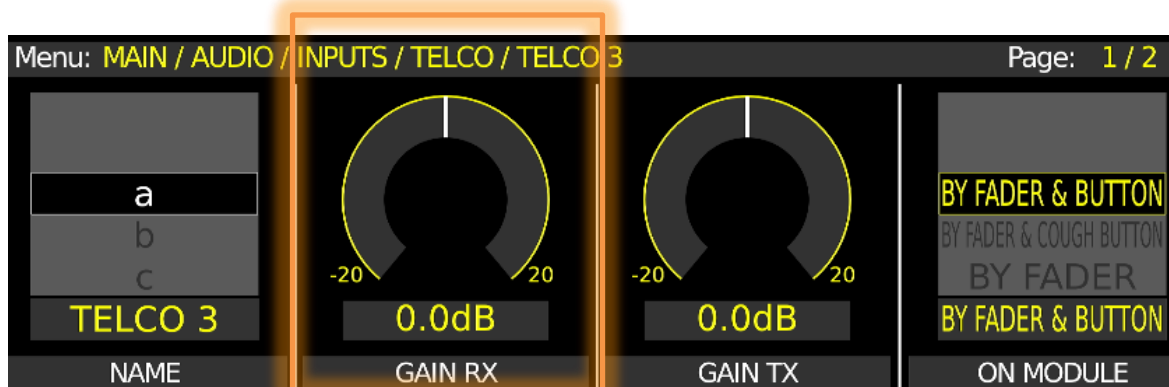
NAME assigns a customised label to the selected TELCO source. The label can contain up to **8 characters**. This limitation applies to the input-source names used by the OXYGEN 3000 Plus.

To edit the TELCO name:

1. Select **NAME**.
2. Rotate the corresponding encoder to select a character.
3. Press the encoder to confirm it.
4. Repeat the operation for the remaining characters.
5. Press **ESC** when the label is complete.

Important: changing NAME does not change the physical connection, external device, GPIO assignment or N-1 routing. It modifies only the label used to identify the source.

GAIN RX:

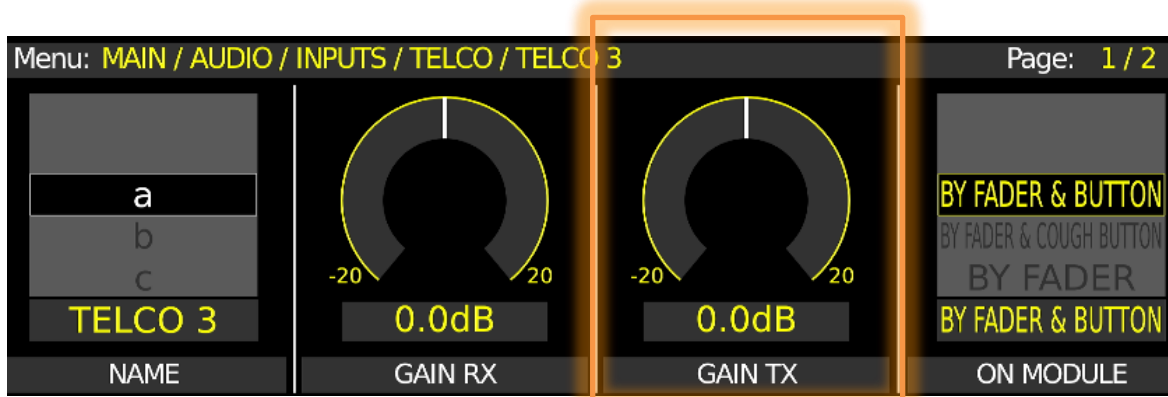


GAIN RX adjusts the level of the audio received from the external telephone hybrid. **Range: -20 dB to +20.0 dB. Adjustment step: 0.1 dB.**

- **Negative values** attenuate the caller signal.
- **0.0 dB** applies unity gain. The caller signal passes through the console without additional digital attenuation or amplification. This is the normal reference setting when the output level of the external hybrid has already been calibrated correctly.
- **Positive values** amplify the caller signal.

Warning: excessive GAIN RX can increase telephone-line noise, background noise and digital artefacts. It can also overload later processing stages.

GAIN TX:

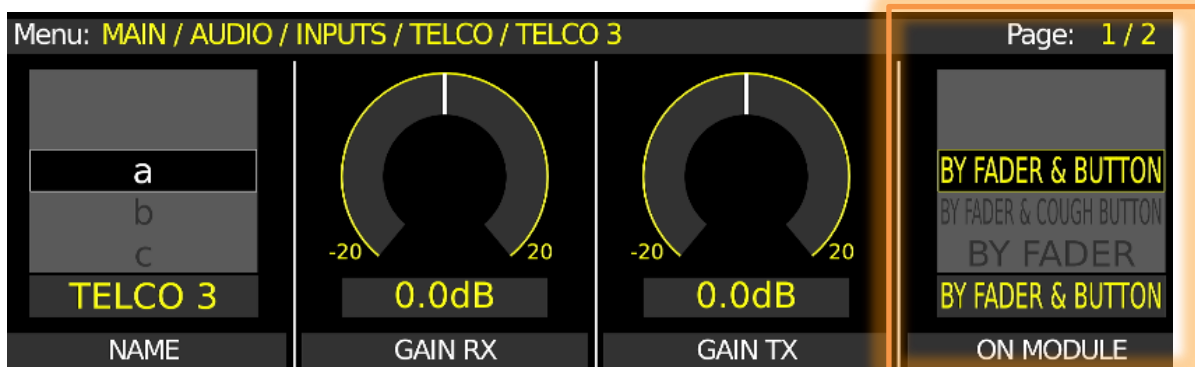


GAIN TX adjusts the level of the audio sent from the OXYGEN 3000 Plus to the external hybrid. **Range:** -20 dB to +20.0 dB. **Adjustment step:** 0.1 dB.

- **Negative values** reduce the level sent to the external hybrid.
- **0.0 dB** passes the N-1 return without additional digital gain. It is the normal reference setting when the console output and external hybrid input use compatible nominal levels.
- Positive values increase the level sent to the external hybrid.

Warning: excessive GAIN TX may overload the external hybrid or reduce the effectiveness of its echo canceller. It may also create distortion in the audio heard by the caller.

ON MODULE:



ON MODULE defines how the channel carrying TELCO changes between its OFF and ON states.

The selectable options are:

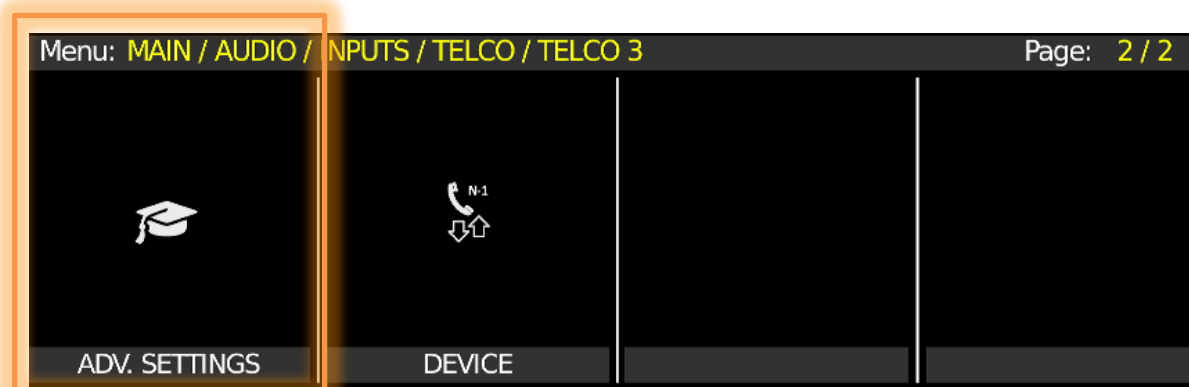
- **FADER & BUTTON:** The TELCO channel is activated through the combined state of: The **physical fader** - The **ON/START button**.

The source becomes active only when the required button state is enabled and the fader is raised above the configured threshold. This mode provides deliberate control and helps prevent a caller from being placed On Air accidentally. **Important:** FADER & BUTTON controls channel activation. The separate F button assigned to TELEPHONE manages RING, HOOK and DROP.

- **BY FADER:** The TELCO channel ON/OFF state follows the physical fader. BY FADER is useful for fast operation in which opening and closing the caller channel must follow the fader movement. **Warning:** raising the fader may immediately put the caller On Air if PGM or another programme bus is already assigned.
- **BY BUTTON:** The TELCO channel is activated through the ON/START button. BY BUTTON is useful when the operator must prepare the level before activating the caller. **Important:** a channel can remain logically ON even when its fader is completely closed. Always check the ON/START illumination.
- **ALWAYS ON:** The TELCO source remains logically enabled. **Warning:** ALWAYS ON may leave caller audio available to pre-fader, direct or auxiliary paths even when the normal channel controls appear inactive.
- **ALWAYS OFF:** The TELCO source remains disabled. The source configuration is retained, but the channel cannot normally be activated from the standard surface controls. **Important:** ALWAYS OFF does not delete the source, DEVICE assignment, GAIN values or advanced settings.

ON MODULE controls the logical activation of the TELCO source. The physical fader continues to determine the audio level whenever it is part of the active signal path.

ADV. SETTINGS:



ADV. SETTINGS opens the additional processing, routing, signalling and surface-control parameters assigned to TELCO. Depending on firmware, the section may include:

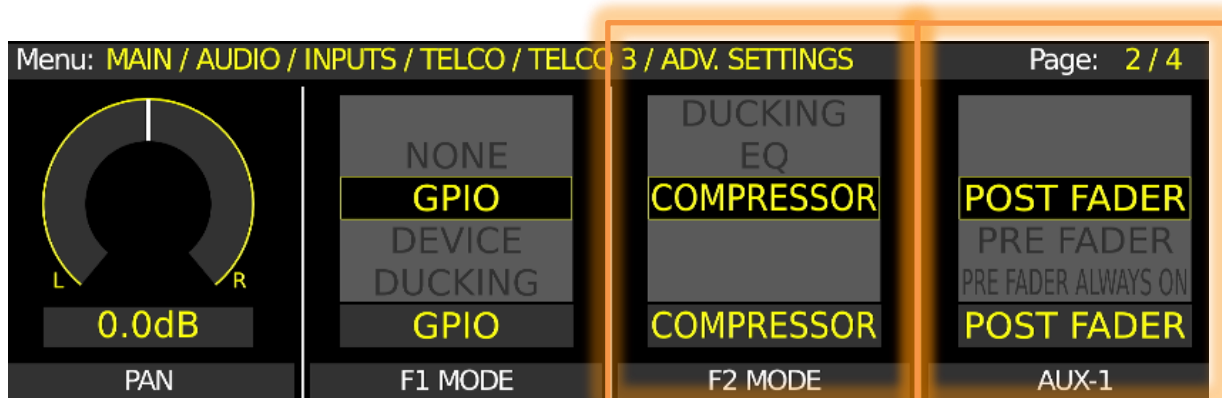
- EQ
- DUCKING
- DYN. COMP.
- MR LIGHT RGB
- PAN
- F1 MODE

- **F2 MODE**
- **AUX-1**
- **AUX-1 PRE-FADER GAIN**
- **AUX-2**
- **AUX-2 PRE-FADER GAIN**
- **BUTTON LIGHT**
- **FADER BAR LIGHT**

Important: ADV. SETTINGS mainly processes and routes the caller's RX signal. GAIN TX and the N-1 output remain separate parts of the return path.

ATTENTION: All the **ADVANCED SETTINGS** listed above are essentially identical to those previously described in the **TELEPHONE** section. The only exceptions are **F1 MODE** and **F2 MODE**, which include additional options described in detail below.

F1 MODE / F2 MODE:



F1 MODE and **F2 MODE** assign functions to the programmable F buttons of the channel carrying TELCO. Depending on firmware, the selectable options include:

- **NONE:** The selected F button performs no action.
- **GPIO:** Assigns GPIO-based control functions to the button, depending on system configuration and external device mapping.
- **DEVICE:** Enables network or device-based control functions for compatible external hybrids or call-management systems –(Only Macrotel X is compatible with this function).
- **DUCKING:** Activates or deactivates ducking when DUCKING STATUS is set to **UNLOCKED**.
- **EQ:** Enables or bypasses the available TELCO equalisation processing.
- **COMPRESSOR:** Activates or deactivates the compressor when its MODE is set to **UNLOCKED**.

Important: assigning a function to F1 or F2 defines the button behaviour. The separate DEVICE parameter defines how the resulting RING, HOOK and DROP commands are exchanged with the external hybrid.

DEVICE:

The **DEVICE** section configures how the OXYGEN 3000 Plus exchanges call-control commands with the external telephone hybrid associated with the selected TELCO source.

Only Macrotel X1/X2 series are compatible with this function.

It can normally be accessed through:

MAIN → AUDIO → INPUTS → PHONE → TELCO → TELCO 1 → DEVICE

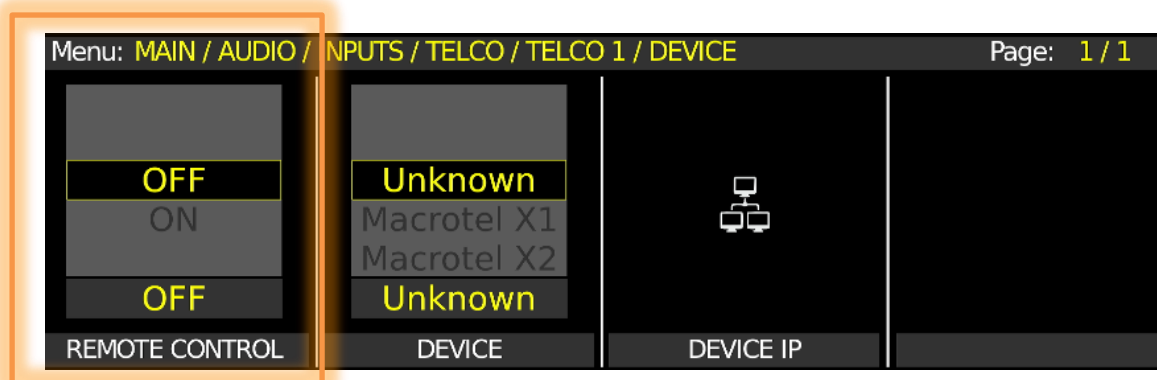
Depending on the installed firmware, the path may instead be displayed as:

MAIN → AUDIO → INPUTS → TELCO → TELCO 1 → DEVICE

The section includes:

- **REMOTE CONTROL:** determines whether remote communication is active.
- **DEVICE:** determines which communication profile must be used.
- **DEVICE IP:** determines where the selected device can be reached on the network.

Important: the DEVICE section configures call signalling and remote control. It does not carry the TELCO RX and TX audio. The external hybrid's audio connections and the corresponding N-1 output must be configured separately.

REMOTE CONTROL:

REMOTE CONTROL determines whether the OXYGEN 3000 Plus exchanges call-status and call-control information with the external telephone hybrid.

The selectable options are:

- **ON:** When **REMOTE CONTROL** is set to **ON**, the OXYGEN 3000 Plus enables call-control communication with the external hybrid. Compatible Macrotel X1/X2 Multimode devices can be

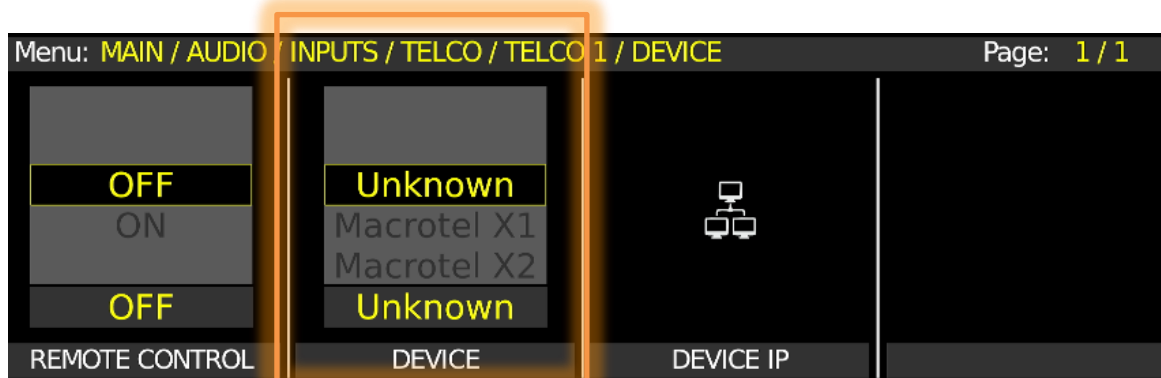
controlled from Oxygen-series consoles through the network by adding the hybrid's IP address.

Important: enabling ON does not send TELCO audio over the network unless a separate network-audio connection has also been configured. This mode normally controls signalling only.

- **OFF:** When REMOTE CONTROL is set to **OFF**, the OXYGEN 3000 Plus does not exchange call-control commands with the external hybrid. **Important:** OFF disables remote call control only. It does not mute the TELCO audio source or remove its N-1 return.

The official OXYGEN 3000 Plus manual identifies these modes for controlling an external hybrid from the channel F button.

DEVICE:



DEVICE selects the external-hybrid communication profile used by the OXYGEN 3000 Plus. A device profile tells the console how to communicate with the selected external hybrid.

When **no device profile** is selected:

- Network commands cannot be correctly formatted
- DEVICE IP may remain stored but is not sufficient by itself
- RING, HOOK and DROP control may not operate
- TELCO audio can remain available

Important: entering DEVICE IP without selecting a compatible DEVICE profile does not complete the network-control configuration.

MACROTEL X1 PROFILE:

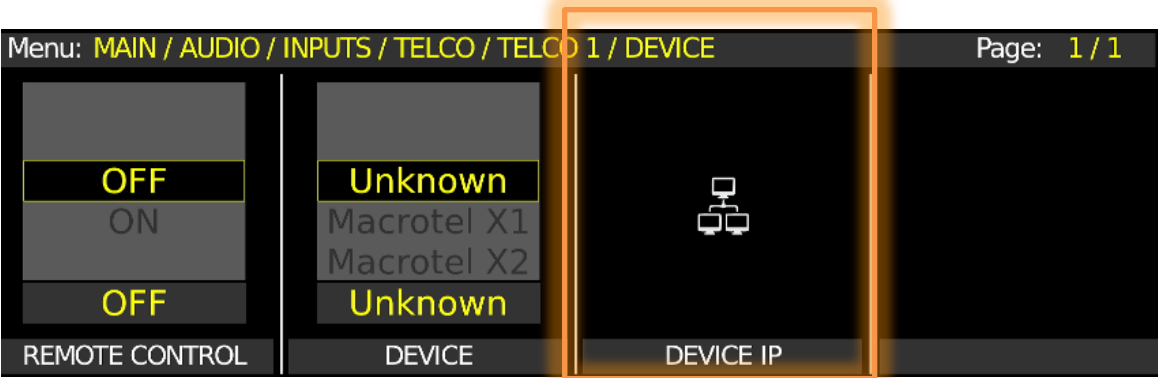
MACROTEL X1 identifies a single-line Macrotel external hybrid. A single-line device normally requires one TELCO source and one corresponding N-1 return.

MACROTEL X2 — LINE 1:

MACROTEL X2 identifies a dual-line Macrotel external hybrid. This selection is used when:

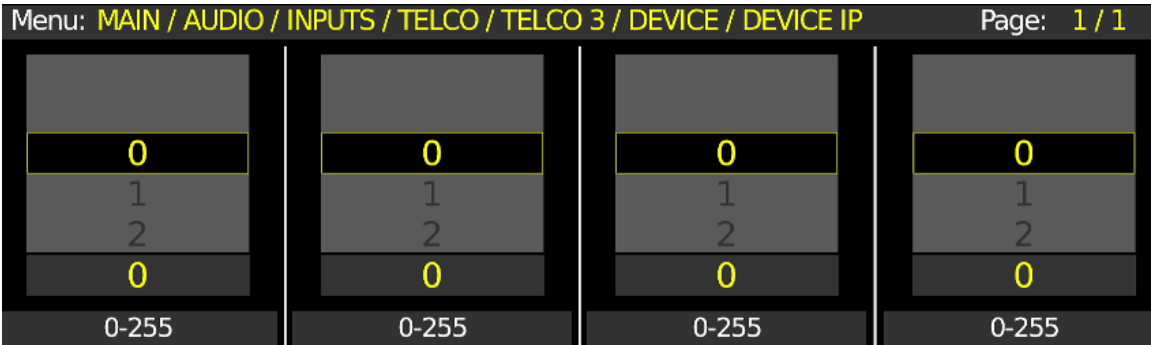
- Two independent telephone lines are available
- Each TELCO source must control a specific line (Line 1 or Line 2)
- Network-based RING, HOOK and DROP control is required

DEVICE IP:



DEVICE IP specifies the IPv4 address of the external hybrid selected under **DEVICE**. It is used only when:

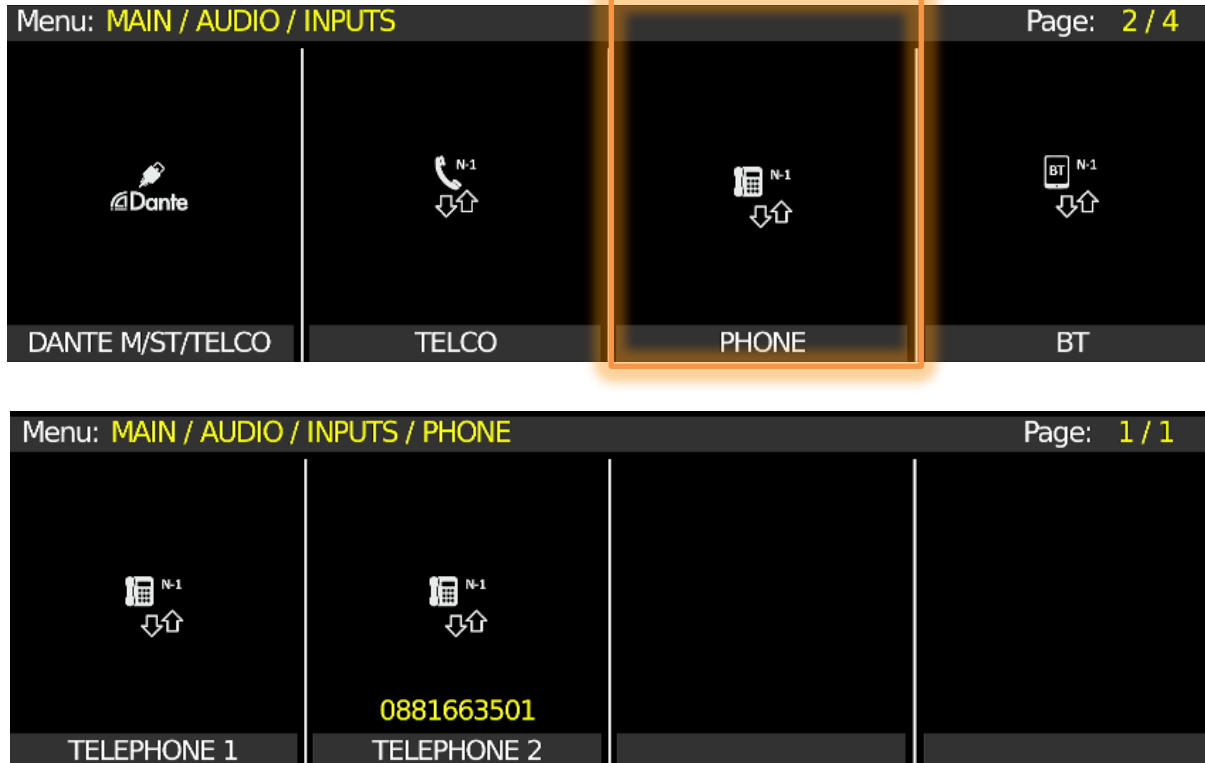
- **REMOTE CONTROL = ON**
- A compatible **DEVICE** profile has been selected
- The external hybrid supports network control



PHONE

The **PHONE** section, located under **AUDIO INPUTS**, configures the audio source associated with the OXYGEN 3000 Plus internal telephone hybrid. It can be accessed through: **MAIN → AUDIO → INPUTS → PHONE → TELEPHONE 1**

When a second compatible Audio I/O Board is installed, an additional telephone resource may be available as: **MAIN → AUDIO → INPUTS → PHONE → TELEPHONE 2**



The PHONE source provides bidirectional communication through an analogue POTS/PSTN telephone line connected to the console's **RJ11 LINE** port.

The telephone connection contains two separate audio paths:

- **RX:** audio received from the remote caller and sent into the console.
- **TX:** audio generated by the console and returned to the remote caller.

The integrated telephone hybrid separates the incoming caller audio from the outgoing return path. The TX return should use the correct N-1 or clean-feed routing so that the caller does not hear their own voice returned with delay. The PHONE connection as an analogue POTS/PSTN telephone-line input on RJ11

Important: PHONE refers to the console's internal telephone hybrid. It must not be confused with TELCO, which is intended for an external telephone hybrid, or BT, which uses a wireless smartphone connection.

ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

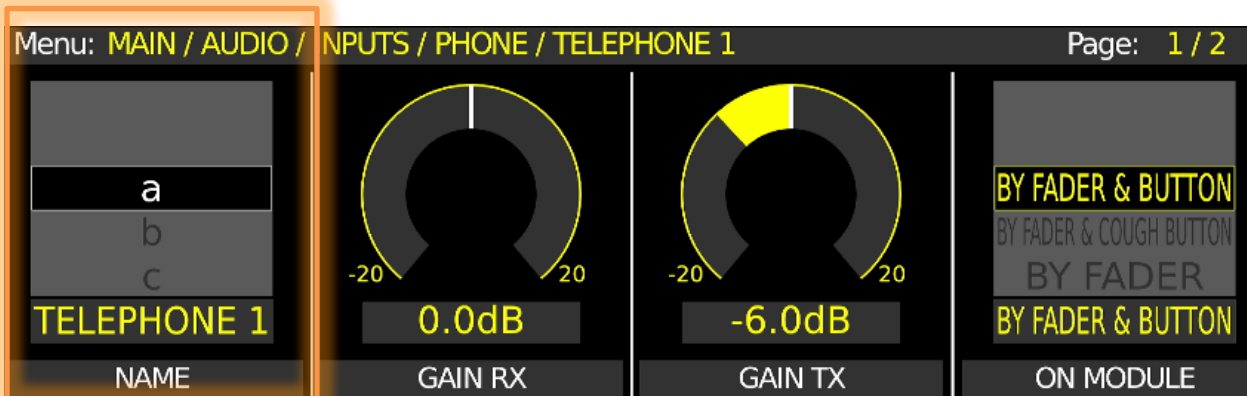
The PHONE page includes:

- **NAME**

- GAIN RX
- GAIN TX
- ON MODULE
- ADV. SETTINGS

INPUTS SETTING (PHONE)

NAME:



NAME assigns a customised label to the selected Phone input. The name can contain up to **8 characters**. Press **ESC** to complete the editing procedure. To edit the label:

1. Rotate the corresponding encoder to select a character.
2. Press the encoder to confirm it.
3. Continue until the required name has been entered.
4. Confirm or exit the editing page according to the displayed controls.

Important: changing NAME does not modify the telephone connection, line number, routing or operating mode. It changes only the label used to identify the source.

GAIN RX:



GAIN RX adjusts the level of the audio received from the remote caller.

The parameter is expressed in decibels and can be adjusted in small steps. Depending on firmware, the available range is normally displayed between negative and positive gain values, with **0.0 dB** representing unity gain. **Range: -20.0 dB to +20 dB. Adjustment step: 0.1 dB.**

- A **negative value** attenuates the incoming caller audio.
- **0.0dB** applies neither attenuation nor additional boost.
- A **positive value** increases the caller level.

Warning: excessive GAIN RX can cause clipping or make telephone-line noise more noticeable. Adjust it while listening through PFL and observing the channel meter.

GAIN TX:

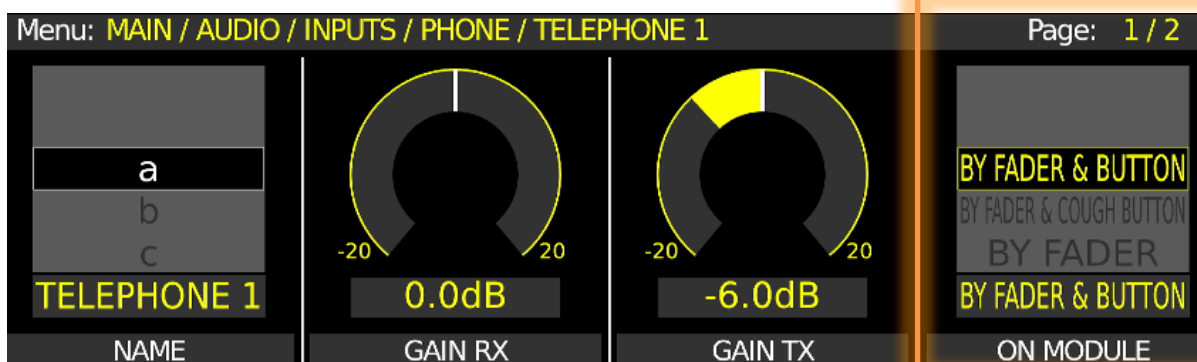


GAIN TX adjusts the level of the audio sent from the console to the remote caller. **Range: -20.0 dB to +20 dB. Adjustment step: 0.1 dB.**

- A **negative value** attenuates the incoming caller audio.
- **0.0dB** applies neither attenuation nor additional boost.
- A **positive value** increases the caller level.

Warning: excessive GAIN TX may cause distortion, echo or uncomfortable listening for the remote caller. Do not use TX gain to compensate for an incorrectly configured N-1 return.

ON MODULE:



ON MODULE determines how the PHONE source changes between its active and inactive audio states when it is assigned to a channel. The selectable options are:

- **FADER & BUTTON:** requires the combined use of: The **physical channel fader** - The **ON/START button**. The PHONE source becomes operational according to both controls and

the configured fader threshold. This mode provides deliberate two-stage activation and can help prevent a caller from being placed On Air accidentally.

- **BY FADER:** When ON MODULE is set to **BY FADER**, the PHONE source's audio state follows the physical channel fader. **In this mode:** Raising the fader above the threshold activates the channel, Lowering the fader below the threshold deactivates it, The ON/START button does not provide the primary activation command, The telephone call remains connected until it is dropped separately. **BY FADER is useful** for rapid broadcast operation after the caller has been checked through PFL. **Warning:** raising the fader may immediately place the caller on the selected buses.
- **BY BUTTON:** When ON MODULE is set to **BY BUTTON**, the PHONE source is activated through the channel's **ON/START** button. **In this mode:** Pressing ON/START activates or deactivates the channel, The physical fader controls the audio level, Moving the fader does not independently change the logical ON/OFF state, HOOK and DROP remain assigned to the telephone-control F button.

This mode keeps line control, channel activation and level control as separate operations.

- **ALWAYS ON:** When ON MODULE is set to **ALWAYS ON**, the PHONE source remains logically active. ALWAYS ON may be useful in specialised workflows where the telephone source must remain continuously available to the routing system. **Warning:** the caller may become audible as soon as the call is answered if the channel fader is raised and a programme bus is already assigned.
- **ALWAYS OFF:** When ON MODULE is set to **ALWAYS OFF**, the PHONE source remains logically inactive. **In this mode:** The source cannot normally be placed On Air from the channel controls, Raising the fader does not activate the source, Pressing ON/START does not activate the source, Stored routing and processing settings remain available, The telephone line may still be answered or managed separately. **Important:** ALWAYS OFF does not delete the PHONE configuration or disconnect the physical telephone line.

Important: ON MODULE controls the PHONE source's audio state on the console. It does not answer or terminate the telephone call. HOOK and DROP are controlled separately through the assigned F button.

ADVANCED SETTING:



The **ADV. SETTINGS** section of TELEPHONE provides the additional processing, routing, signalling and surface-control parameters associated with the internal telephone-hybrid source.

It can be accessed through:

MAIN → AUDIO → INPUTS → PHONE → TELEPHONE 1 → ADV. SETTINGS

or, when a second telephone resource is available:

MAIN → AUDIO → INPUTS → PHONE → TELEPHONE 2 → ADV. SETTINGS

Depending on the installed firmware and available functions, the section may include:

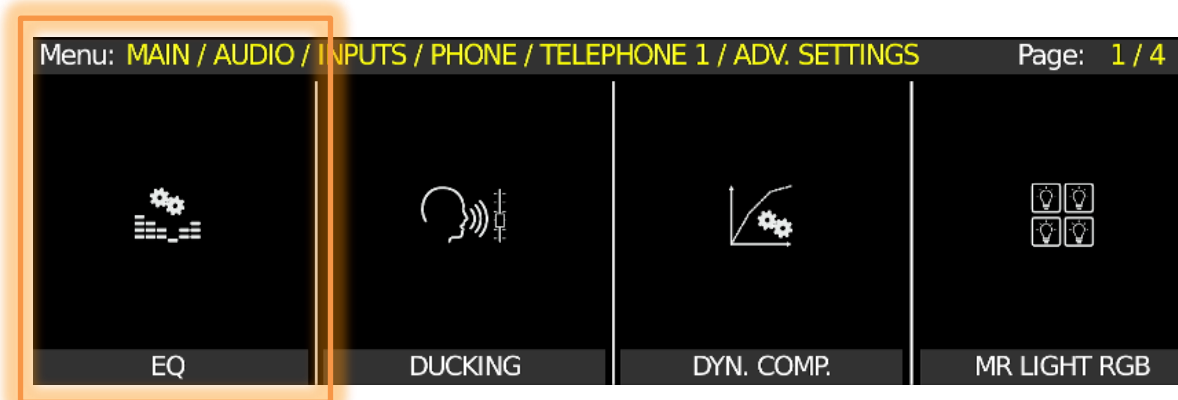
- EQ
- DYN. COMP.
- DUCKING
- MR LIGHT RGB
- PAN
- F1 MODE
- F2 MODE
- AUX 1
- AUX 1 PRE-FADER GAIN
- AUX 2
- AUX 2 PRE-FADER GAIN
- BUTTON LIGHT
- FADER BAR LIGHT

The telephone-specific F-button assignment can be accessed through:

MAIN → AUDIO → INPUTS → PHONE → TELEPHONE 1 → ADV. SETTINGS

It assigns the required HOOK and DROP operation to the channel carrying the TELEPHONE source. This assignment does not automatically control TELCO or BT sources.

EQ:



EQ opens the equalizer assigned to the PHONE receive signal. It may be used to:

- Reduce low-frequency line hum
- Control excessive bass
- Improve voice presence

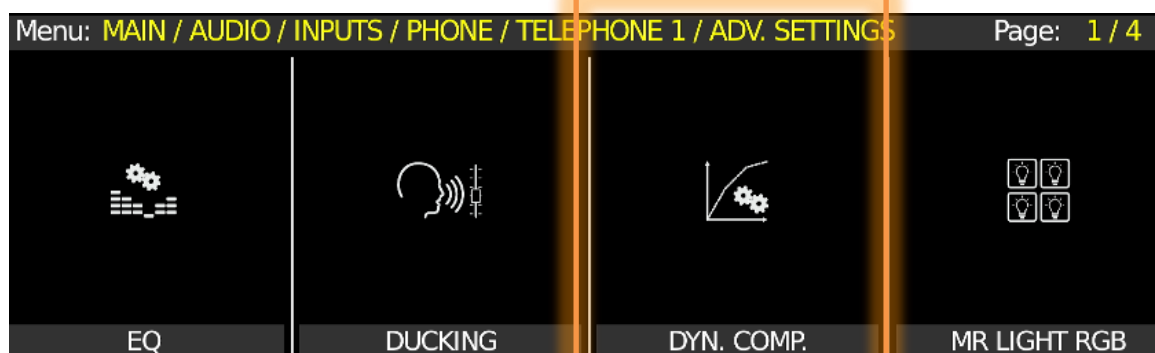
Telephone audio has a restricted bandwidth, so excessive boosts should be avoided.

ATTENTION: A detailed description of this mode can be found in the section dedicated to the MIC input

DUCKING:

DUCKING configures how the TELEPHONE source interacts with the console's automatic attenuation system. Ducking temporarily lowers one or more slave sources when a configured master source becomes active. It is commonly used to reduce music or another background source while a presenter, guest or caller is speaking. For TELEPHONE, ducking may be used in Slave mode: The caller can operate as a **SLAVE**, being attenuated when another master becomes active.

ATTENTION: A detailed description of this mode can be found in the section dedicated to the MIC input

DYN. COMP.:

DYN. COMP. opens the dynamics-processing section assigned to the TELEPHONE receive signal.

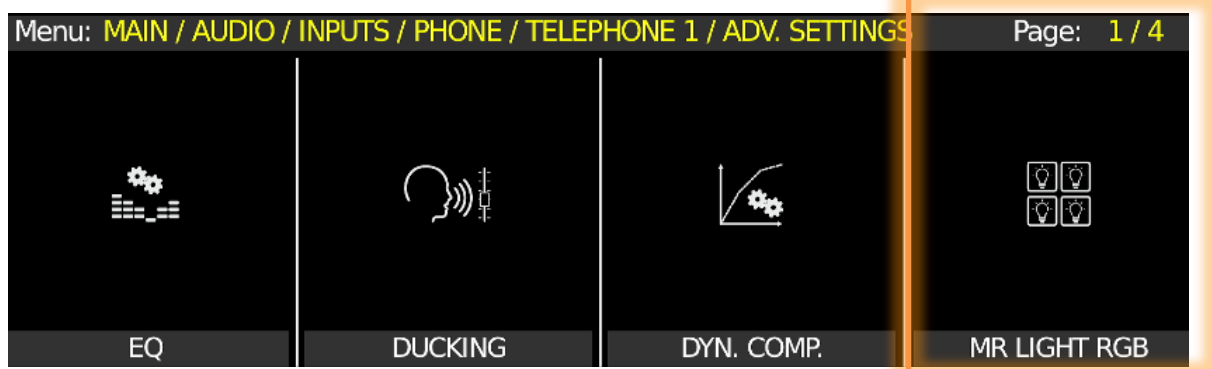
Telephone levels can vary considerably between different callers and telephone networks. Dynamics processing can help:

- Control sudden caller peaks
- Improve level consistency
- Reduce low-level telephone-line noise
- Prevent excessive signal levels
- Improve speech intelligibility

Important: DYN. COMP. processes the caller's audio inside the console. It must not be used as a replacement for correct GAIN RX adjustment.

ATTENTION: A detailed description of this mode can be found in the section dedicated to the MIC input

MR LIGHT RGB:

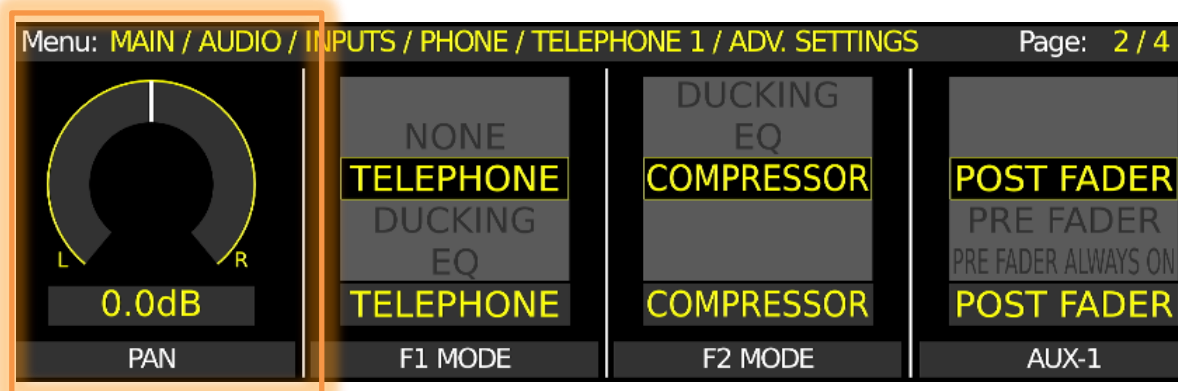


MR LIGHT RGB associates the TELEPHONE source with a compatible Axel Technology MR LIGHT RGB signalling lamp. MR LIGHT RGB supports configurable colours, brightness, graphics and operating modes. The selectable values normally include:

- **OFF** or **NONE**: No MR LIGHT RGB device is associated with TELEPHONE. Changes to the TELEPHONE channel do not generate a command for an external MR LIGHT RGB lamp.
- A configured **MR LIGHT RGB device**: Associates TELEPHONE with the selected lamp. The displayed choices depend on the devices configured and detected by the system.
- A logical device name or identifier, depending on system configuration

ATTENTION: For further information about the configurable options, refer to the corresponding section under **MIC** or **MONO**.

PAN:



PAN positions the mono TELEPHONE receive signal between the Left and Right channels of stereo buses. The available adjustment extends continuously from:

- Full **LEFT**: The caller is positioned mainly or completely in the Left channel.
- Through the centre position: The caller is distributed equally between the Left and Right channels. This is the normal setting for most telephone calls.
- To full **RIGHT**: The caller is positioned mainly or completely in the Right channel.

The parameter is adjusted in **1.0 dB steps**.

Important: PAN affects the caller's RX position in stereo buses. It does not affect GAIN TX or the mono return sent to the caller.

F1 MODE / F2 MODE:

F1 MODE and **F2 MODE** assign functions to the two programmable F buttons of the channel carrying the TELEPHONE source. The selectable options are:

- **NONE:** The selected F button performs no action. Pressing it does not change call control, processing or communication status.
- **TELEPHONE:** Assigns the internal telephone-hybrid control to the selected F button.
The button normally manages: - **Incoming-call** indication - **HOOK** - **DROP**
The usual LED states are: **LED off:** telephone line not hooked - **LED flashing:** incoming call - **LED on:** telephone line hooked
Pressing the assigned button answers the incoming call. Pressing it again drops the connected call.
- **DUCKING:** The selected F button activates or deactivates the TELEPHONE ducking function. This control is available only when DUCKING STATUS is set to **UNLOCKED**.
- **EQ:** The selected F button enables or bypasses the equalisation processing associated with the TELEPHONE channel (when available in the system configuration). This allows quick comparison between processed and unprocessed caller audio.
- **COMPRESSOR:** The selected F button activates or deactivates the TELEPHONE compressor. This control is available only when COMPRESSOR MODE is set to **UNLOCKED**.

The options available for F1 and F2 are configured independently.

Important: for normal internal-hybrid call management, at least one F button must be assigned to TELEPHONE.

AUX-1 / AUX-2:



AUX-1 and **AUX-2** independently determine how the TELEPHONE receive signal is sent to the corresponding auxiliary bus.

The selectable options are:

- **POST FADER:** The TELEPHONE contribution to the AUX bus follows the physical channel fader. POST FADER is useful when the AUX output must follow the On Air caller level.
- **PRE-FADER:** The TELEPHONE contribution is taken before the physical channel fader. PRE-FADER can be useful for: Recording feeds - Independent monitoring - Communication buses - Production mixes.
- **PRE-FADER ALWAYS ON:** The TELEPHONE contribution is independent from: The physical fader - The ON/START button
The caller can therefore remain present in the selected AUX bus even when the channel is OFF and its fader is completely closed.

Warning: PRE-FADER ALWAYS ON can leave a telephone conversation audible on an AUX output without an obvious indication from the channel fader. Verify privacy and routing before using this mode.

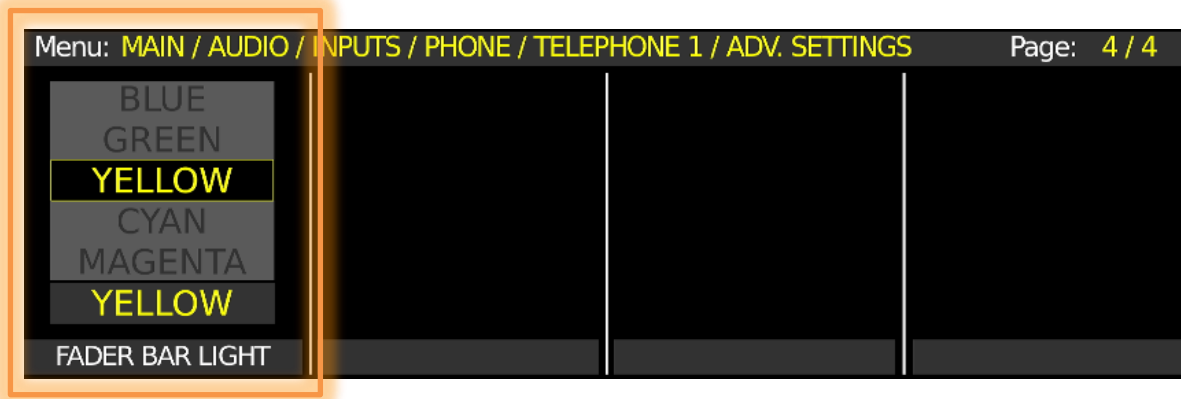
AUX-1 PRE-FADER GAIN / AUX-2 PRE-FADER GAIN:

These parameters independently set the TELEPHONE receive level sent to the AUX-1 and AUX-2 pre-fader paths. **Range:** -INF to +10 dB. **Adjustment step:** 1.0 dB

Important: AUX-1 PRE-FADER GAIN and AUX-2 PRE-FADER GAIN are independent. Muting one pre-fader path does not mute the other AUX bus or the main TELEPHONE channel.

BUTTON LIGHT:

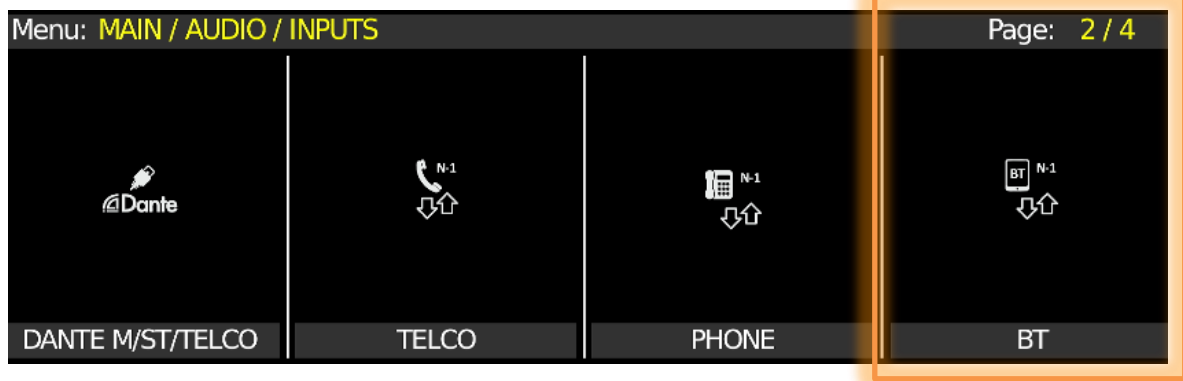
BUTTON LIGHT selects the colour preset used by the illuminated buttons of the physical or virtual channel carrying TELEPHONE. **Important:** BUTTON LIGHT selects a colour preset. It does not change the function assigned to the button or the meaning of its flashing and steady illumination states.

FADER BAR LIGHT:

FADER BAR LIGHT selects the colour preset used by the LED bar associated with the TELEPHONE channel fader. The fader-bar colour provides a persistent visual identification of the source assigned to the channel. **Important:** BUTTON LIGHT and FADER BAR LIGHT are independent. The buttons and fader bar can use the same colour or two different presets.

BT - BlueTooth

The **BT source** is the audio input/output associated with the Bluetooth module integrated into the **AxelTech Oxygen 3000 Plus**. It allows a smartphone, tablet or compatible computer to exchange audio with the console through a wireless connection, without requiring an external analogue or digital audio cable. Bluetooth can be used to put on air **regular mobile-phone calls**, calls made through applications such as **Skype, Microsoft Teams and WhatsApp**, or audio content played through **YouTube, music players and other multimedia applications**.



The Oxygen 3000 Plus provides **one Bluetooth module for each Audio I/O Board**. A console equipped with one Audio Core therefore provides **BT-1**, while a two-core configuration can provide both **BT-1 and BT-2**. The console uses an **BM64 Bluetooth audio module**. Although BT is represented in the console's input/output block diagram, it does not correspond to a dedicated XLR or RJ45 audio connector. The physical connection is the **Bluetooth wireless radio link**, while the received and transmitted signals are managed internally by the console's audio matrix.

In the console block diagram, **BT-1** and **BT-2** are shown as internal input/output paths associated with the first and second Audio Core. BT-1 is represented next to **LINE-ST4**, while BT-2 is represented next to **LINE-ST8**. Their effective availability and internal allocation may depend on the installed hardware and the selected **Input Mode configuration**. The diagram should therefore be interpreted as an internal routing representation and not as a separate physical audio connector or as an unconditional indication that the corresponding line input is disabled.

The Oxygen 3000 Plus supports **two Bluetooth connection modes**. The active profile is normally determined by the paired device and by the type of audio operation being performed.

In **Mono Bidirectional mode**, Bluetooth is used for telephone calls. The caller's mono audio signal travels from the smartphone or tablet to the console through the **BT input path**. At the same time, the console can send a mono return signal to the paired device through the **BT output path**. The return can operate according to **N-1, or clean-feed, logic**: the caller receives the required microphones and programme sources, while the caller's own BT input is excluded from the return. This prevents the caller's voice from being sent back to the same device and helps reduce **echo and acoustic feedback**. The exact content of the return signal depends on the telephone and routing configuration applied to the console.

The bidirectional signal flow can be represented as follows:

Caller or communication application → BT mono input → assigned mixer channel → selected audio buses
Configured microphones and programme sources → N-1 return → BT mono output → paired device

In **Stereo Monodirectional mode**, Bluetooth uses the **A2DP profile** to receive high-quality stereo audio from a smartphone or tablet. This mode is suitable for putting music, recorded programmes, internet audio, YouTube content or media-player audio on air. The audio travels only **from the Bluetooth device to the console**; the BT return path is not used for programme audio in this mode.

The A2DP signal flow is therefore:

Smartphone or tablet → BT stereo input → assigned mixer channel → selected audio buses

To pair a device, briefly press the console's **BT button for less than one second**. The button begins **flashing blue**, indicating that the Bluetooth module is in pairing mode. Open the Bluetooth settings on the smartphone, tablet or computer, search for **Oxygen XXXX**, and select it. Once the connection has been established, the blue indicator stops flashing and remains **continuously illuminated**.

After pairing, rotate or press the encoder of the required mixer channel and search for **BT** in the source-selection menu—for example, the encoder associated with Channel 4. The BT source can then be assigned to the required channel position. Once the source has been selected, press the desired bus button, such as **PGM**, to route the Bluetooth signal to that bus. Finally, open the channel according to its **ON MODULE** configuration by using the **ON/START** button, the fader or both controls.

The physical channel controls manage the BT source in the same way as the other mixer inputs. The **GAIN encoder** adjusts the level received from the paired device, while the **fader** determines the level sent to post-fader destinations. The **ON/START button** opens or closes the channel, and the **PGM, SUB, AUX 1 and AUX 2 buttons** select the buses to which the Bluetooth audio is routed.

Once included in a bus, the BT signal can reach any console output assigned to that bus. For example, when the source is routed to **PGM**, it becomes part of the programme mix and may consequently be distributed through the configured **analogue outputs, AES/EBU output, USB interfaces, Dante outputs or streaming encoder**. The Bluetooth signal is therefore not connected directly to a specific physical output: it first passes through the input channel, fader, audio matrix and bus-routing system.

During a telephone call, the **PFL button** can be used for private off-air communication with the caller. When PFL is activated on the BT channel, microphones configured as **PRIVATE MIC** can communicate with the caller independently of their normal fader and ON/START status. The private conversation remains excluded from the main programme outputs, allowing the presenter or technician to speak with the caller before putting the call on air.

Depending on the selected Bluetooth profile, paired device and installed console firmware, an **F1 or F2 button** may also be configured to display or control the telephone connection status. Because this behaviour can vary according to the connected device and firmware version, its operation should be verified during system configuration before being described as a fixed call-answering or call-ending control.

Once the BT source has been assigned and routed, start the required audio stream—such as music, a YouTube video or a media-player track—or begin the regular telephone, Skype, Teams or WhatsApp call

from the paired device. If Bluetooth is subsequently enabled again and the device remains stored in the console's pairing memory, the connection is normally restored automatically. A **fixed blue indicator** confirms that the device is connected.

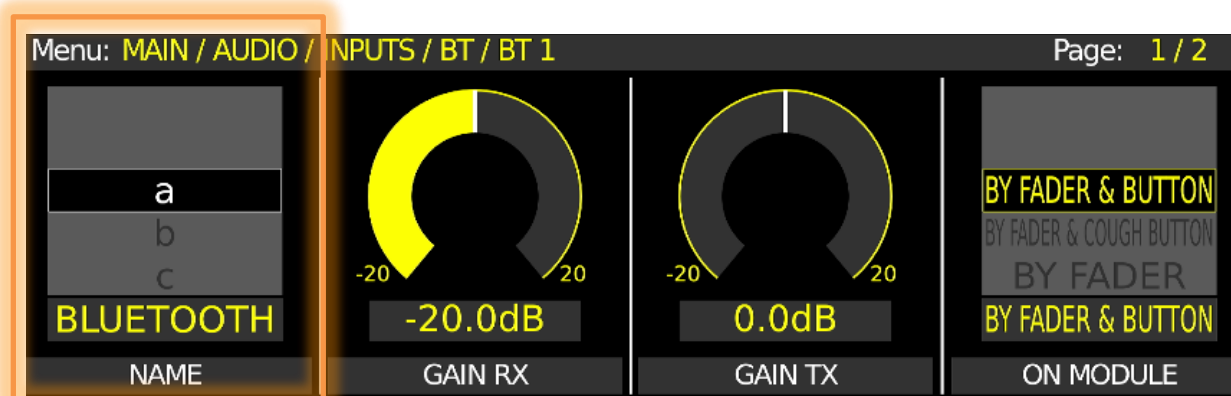
The BT module therefore provides two main functions: **wireless stereo playback through A2DP** and **mono bidirectional communication for telephone calls**, including the possibility of private caller communication and an appropriately configured N-1 return.

ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

INPUTS SETTING (BT)

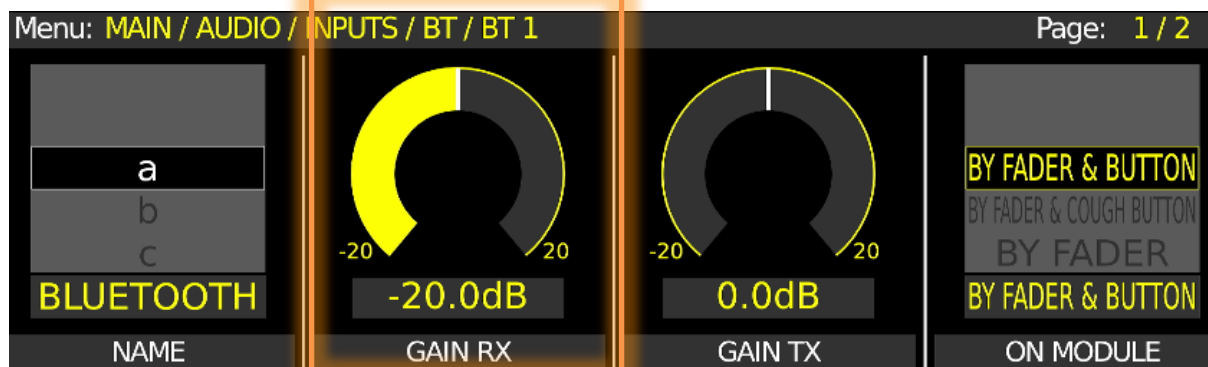
The **BT settings** control the name, incoming and outgoing audio levels, channel activation logic and advanced routing parameters of the Bluetooth source. The Oxygen 3000 Plus provides one BT source for each installed Audio I/O Board: **BT-1** for the first Audio Core and **BT-2** for the second Audio Core, where available.

NAME:



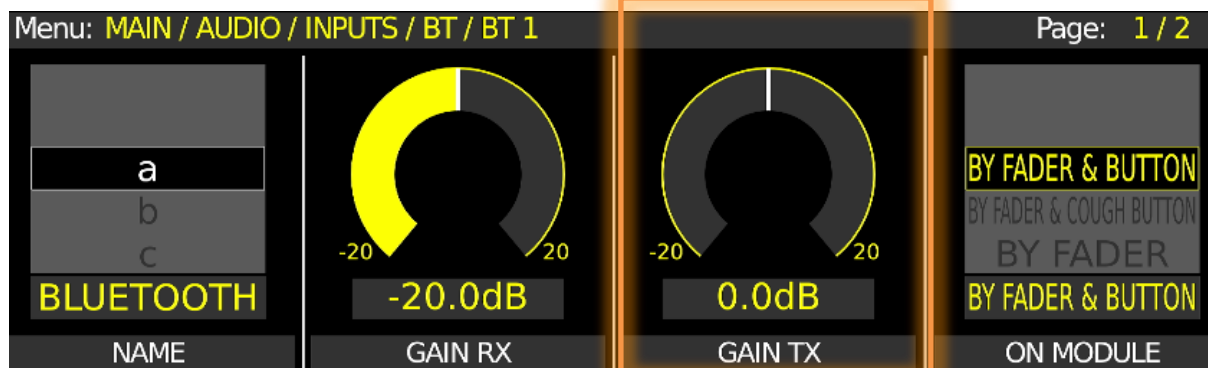
The **Name** parameter assigns a customised label to the Bluetooth source. The selected name is displayed whenever BT is assigned to a physical or virtual mixer channel, allowing the operator to identify it quickly. The label can contain up to **8 characters**. Typical names include **BT-1**, **MOBILE**, **PHONE** or **MUSIC**.

GAIN RX:



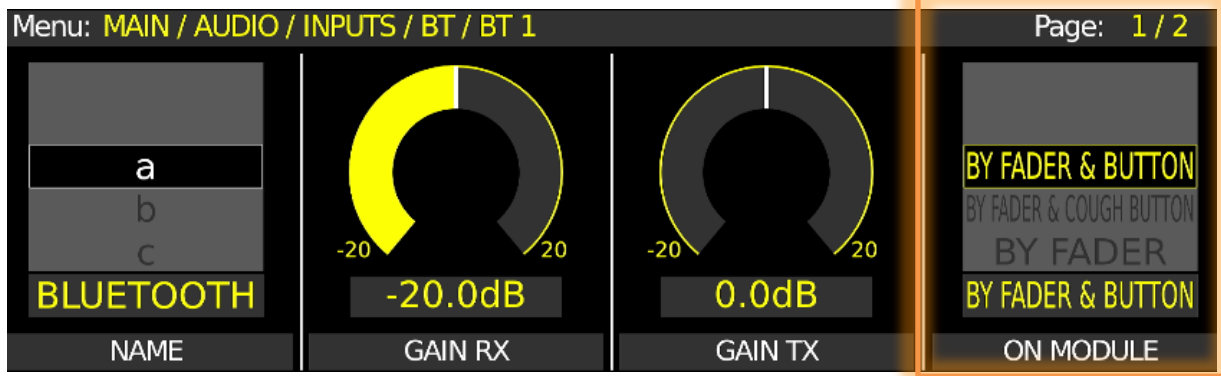
The **Gain RX** parameter adjusts the level of the audio **received by the console from the paired Bluetooth device**. In **Mono Bidirectional mode**, it controls the level of the caller's voice received from a telephone or communication application. In **Stereo A2DP mode**, it controls the level of the stereo music or multimedia content received from the smartphone, tablet or computer. **Range: -20.0 dB to 20.0 dB. Adjustment step: 0.1 dB.** A negative value attenuates an excessively loud Bluetooth signal, while a positive value increases a low incoming level. **Gain RX should be adjusted while monitoring the pre-fader meter**, avoiding signal clipping.

GAIN TX:



The **Gain TX** parameter adjusts the level of the audio **transmitted by the console to the paired Bluetooth device**. This setting is used primarily in **Mono Bidirectional telephone mode**, where the caller must receive the presenter's microphone and the configured programme or N-1 return. It also affects the level sent during a private conversation activated through the BT channel's **PFL function**. **Range: -20.0 dB to 20.0 dB. Adjustment step: 0.1 dB** Increasing Gain TX makes the console return louder for the caller, while decreasing it prevents distortion or an excessively high listening level. **Gain TX does not control the caller's level on air;** that level is controlled by **Gain RX and the channel fader**. In **Stereo Monodirectional A2DP mode**, the Bluetooth connection carries audio only from the paired device to the console. Therefore, **Gain TX has no practical effect on the A2DP programme stream**.

ON MODULE:

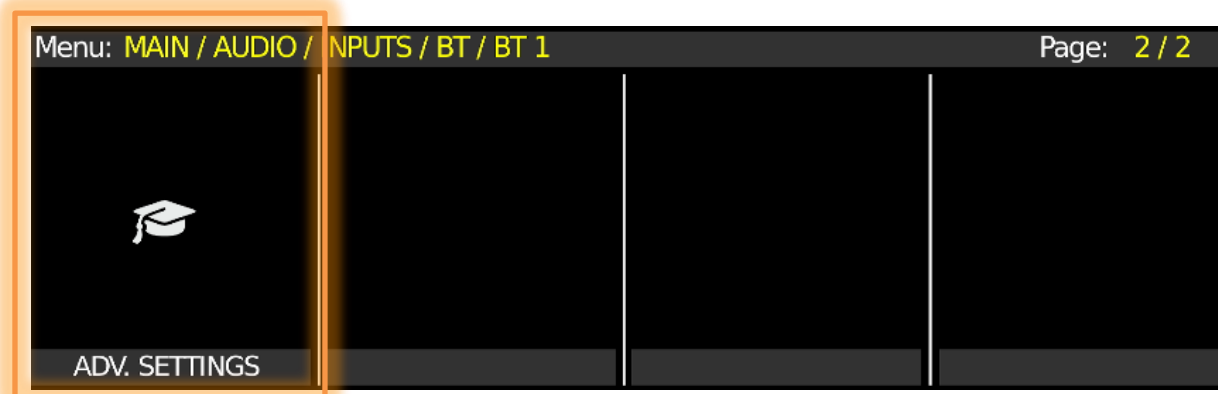


The **On Module** parameter determines how the BT source is opened, closed or temporarily muted from the mixer channel to which it is assigned. The available options are:

- **FADER & BUTTON:** the channel is enabled through the combined operation of the **ON/START button and fader**. The button enables the channel, while the fader controls its opening and audio level.
- **BY FADER:** raising the fader above the configured activation threshold opens the channel automatically. Lowering it below the threshold closes the channel.
- **BY BUTTON:** the channel is opened or closed through the **ON/START button**. The fader controls the audio level but does not determine the channel's ON/OFF status.
- **BY FADER & COUGH BUTTON:** the fader controls the normal opening and closing of the channel, while the configured **COUGH button temporarily mutes the source**. When the Cough button is released, the audio is restored automatically, provided that the fader remains above the activation threshold. This function does not disconnect or unpair the Bluetooth device; it only acts on the assigned mixer channel. Its effect on the BT return signal may depend on the configured routing.
- **ALWAYS ON:** the BT source remains enabled independently of the ON/START button. The fader can still control the level sent to post-fader buses.
- **ALWAYS OFF:** the BT source remains disabled and cannot be opened using the channel controls.

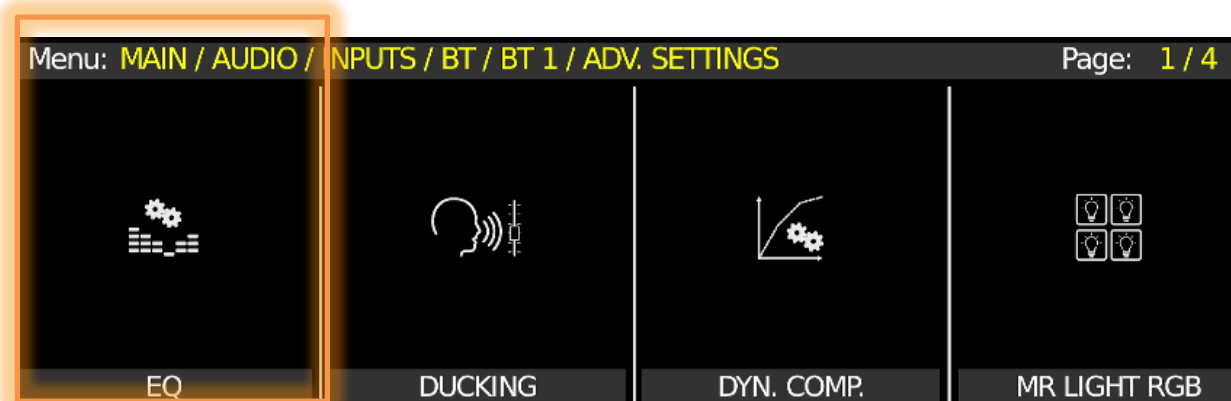
For telephone calls, **FADER & BUTTON** or **BY BUTTON** generally provides the safest operation, because it reduces the possibility of putting a caller on air accidentally.

ADV. SETTINGS:



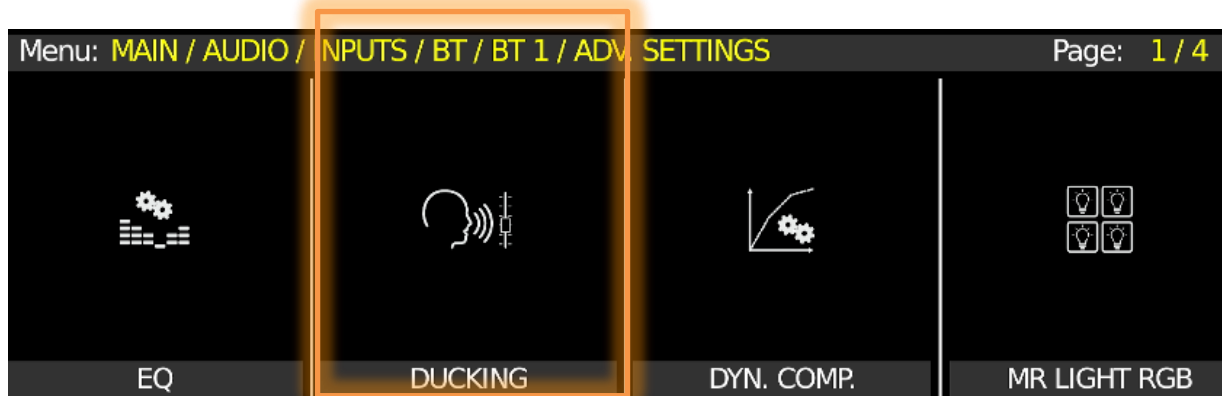
The **Advanced Settings** menu provides access to the Bluetooth source's processing, routing, programmable-button and visual-identification parameters.

EQ:



The **EQ** section provides a multi-band parametric equalizer for adjusting the tonal balance of the received Bluetooth signal. It can be used to improve speech intelligibility during telephone calls or correct the frequency balance of music received through A2DP.

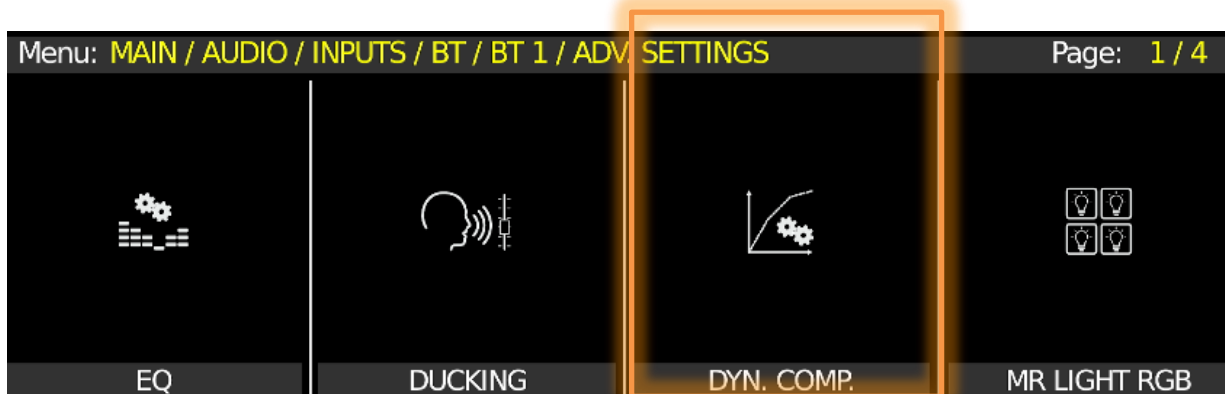
ATTENTION: A detailed description of this mode can be found in the section dedicated to the MIC input

DUCKING:

The **Ducking** section allows the BT source to participate in the console's automatic attenuation system.

SLAVE MODE: enables or disables ducking on the BT source. When enabled, the BT level is automatically reduced whenever an assigned master source becomes active. This is particularly useful when BT is used for **A2DP music playback** and must be lowered while a presenter speaks.

- **ON:** the BT source operates as a ducking slave.
- **OFF:** automatic attenuation is disabled for the BT source.

DYN. COMP.:

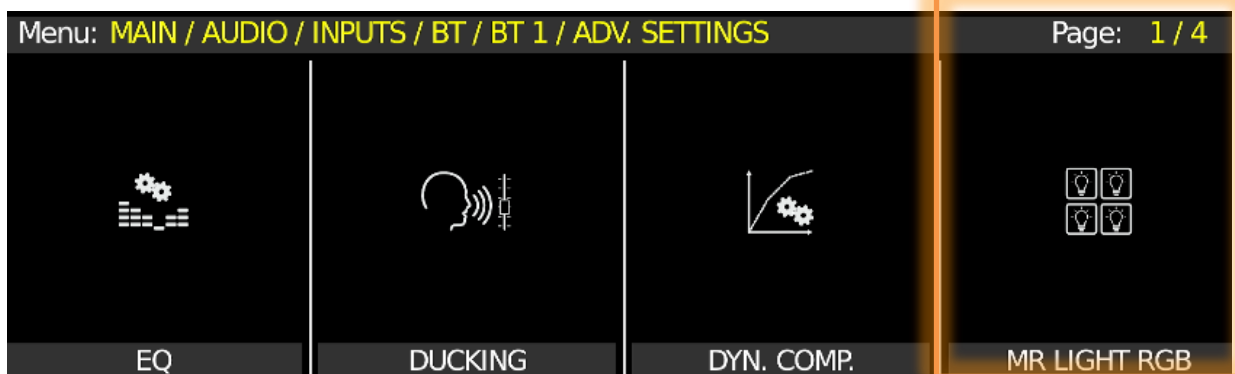
The **DYN. COMP.** section provides dynamic processing for the BT input. It can reduce excessive level variations during telephone calls, control peaks in streamed audio and attenuate background noise when the received signal is inactive.

The section includes:

- **DYNAMIC PRESET:** saves or recalls a predefined set of compressor and noise-gate parameters.
- **COMPRESSOR:** reduces the level of the BT signal when it exceeds the configured threshold. Its parameters include **Compression Ratio, Compression Threshold, Limiting Threshold, Attack Speed, Release Speed, Hold Time** and **Makeup Gain**.

ATTENTION: A detailed description of this mode can be found in the section dedicated to the MIC input

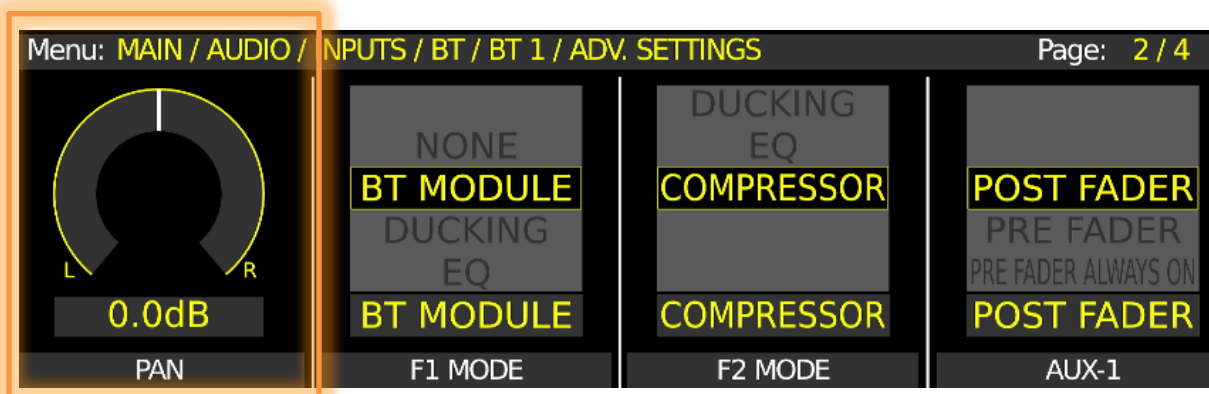
MR LIGHT RGB:



The **MR LIGHT RGB** function associates the BT source with an AxelTech **MR. LIGHT RGB** studio signalling lamp. It allows the lamp to react when the BT source changes status—for example, when a caller or Bluetooth contribution is put on air.

ATTENTION: For further information about the configurable options, refer to the corresponding section under **MIC** or **MONO**.

PAN:

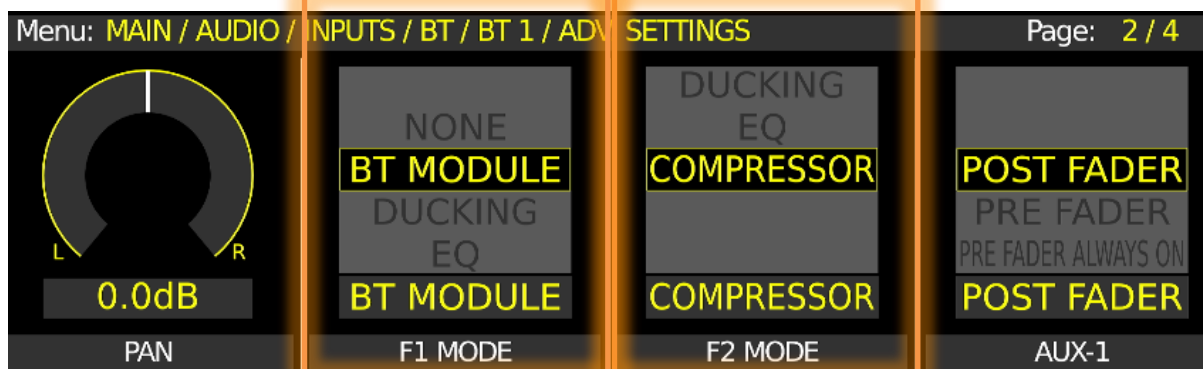


the **Pan** parameter determines the position of the BT source within the stereo output buses.

The available positions range from:

- **LEFT:** moves the source towards the Left output channel.
- **CENTER:** sends the source equally to the Left and Right channels.
- **RIGHT:** moves the source towards the Right output channel.

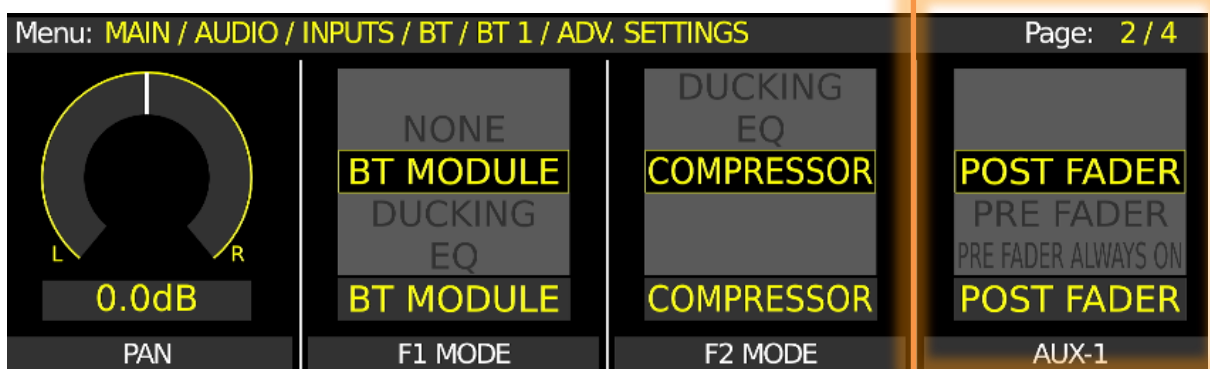
When BT is operating as a mono telephone source, PAN places the caller at the desired position in the stereo mix. When BT is receiving an A2DP stereo stream, the control should normally remain centred unless a specific routing correction is required.

F1 MODE / F2 MODE:

The **F1 Mode** and **F2 Mode** parameters assign a function to the two programmable buttons associated with the BT source. The selectable options include:

- **NONE:** no function is assigned.
- **BT MODULE:** associates the F button with the Bluetooth telephone function. The call is normally answered on the paired device; where supported by the Bluetooth profile, device and console firmware, the F button can forward the connected call to the BT channel or terminate it.
- **EQ:** enables or disables the BT equalizer.
- **COMPRESSOR:** enables or disables the compressor.
- **DUCKING:** controls or indicates the permitted ducking function.

A function configured as **LOCKED ON** or **LOCKED OFF** cannot be overridden by its assigned F button. The exact options displayed on the console must always be considered authoritative for the installed firmware.

AUX 1 / AUX 2:

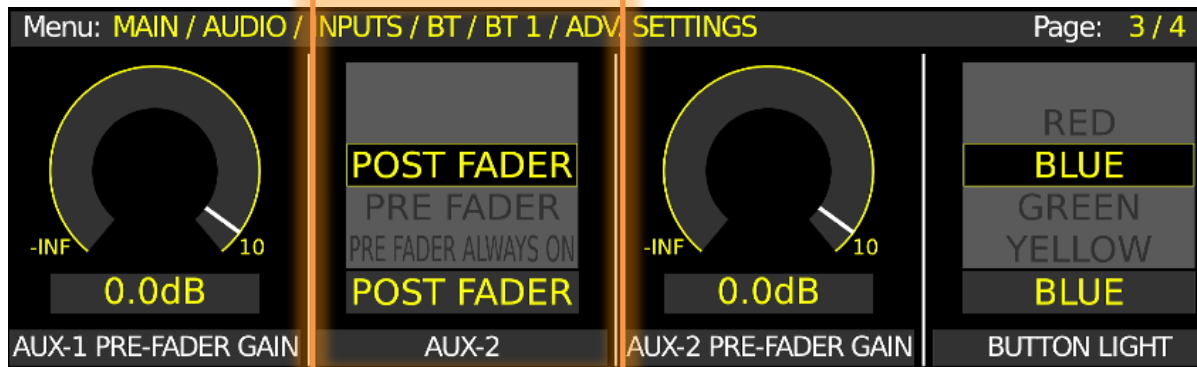
The **AUX 1** and **AUX 2** parameters determine where the BT signal is taken in relation to the channel fader and ON/START status before being sent to the corresponding auxiliary bus.

The selectable modes are:

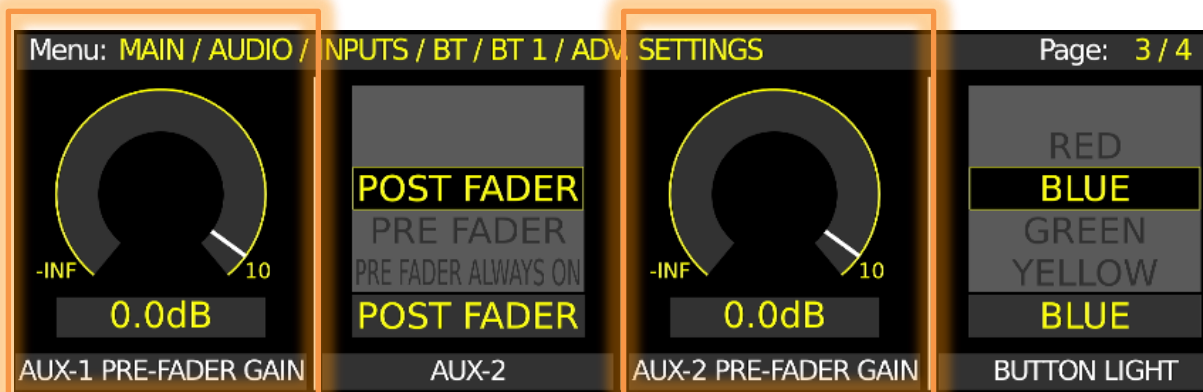
- **POST-FADER:** the AUX signal follows the BT channel fader and channel activation status.
- **PRE-FADER:** the AUX signal is independent of the channel fader but remains controlled by the channel's ON/OFF status.

- **PRE-FADER ALWAYS ON:** the AUX signal is independent of both the fader and ON/START button and remains continuously available on the selected auxiliary bus.

POST-FADER is suitable when the auxiliary level must follow the on-air mix. **PRE-FADER** is suitable for independent monitoring or recording, while **PRE-FADER ALWAYS ON** can be used for permanently available service or communication feeds.



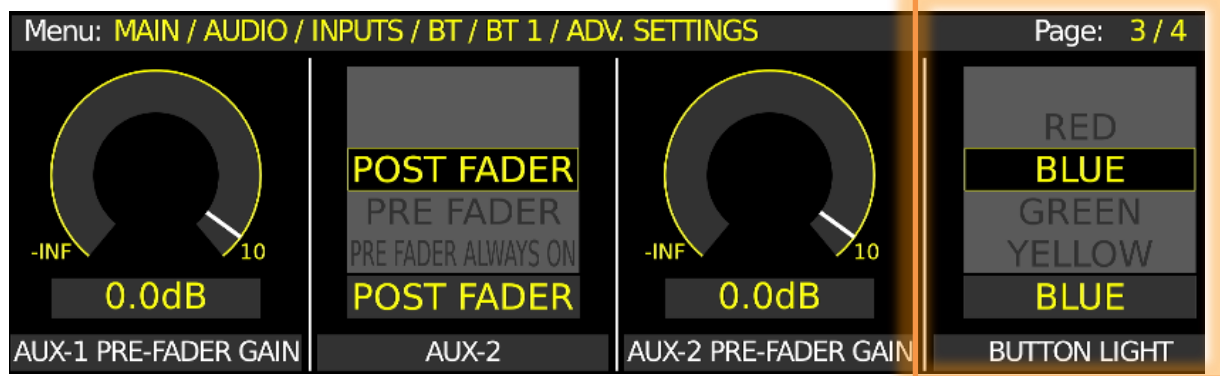
AUX 1 PRE-FADER GAIN / AUX 2 PRE-FADER GAIN:



These parameters adjust the BT signal level sent to **AUX 1** or **AUX 2** when the corresponding bus is operating in a pre-fader mode.

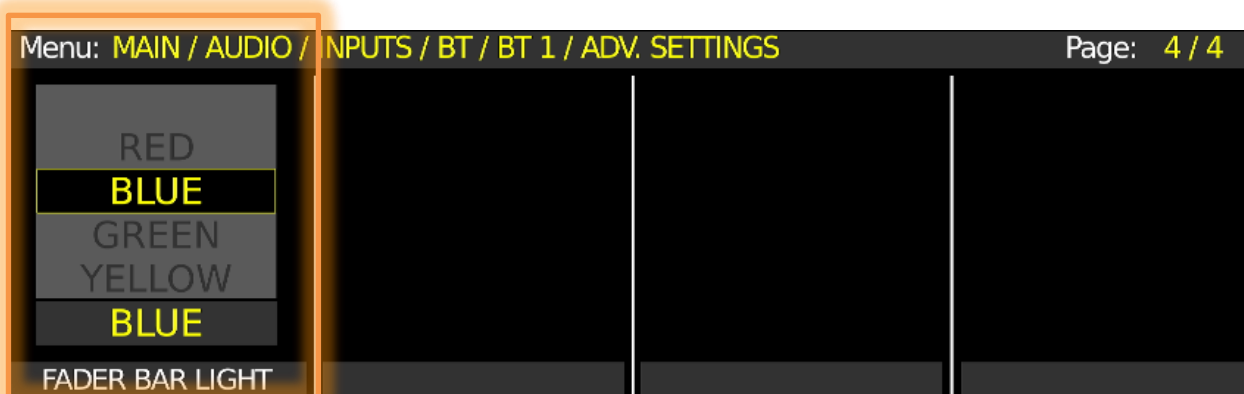
- A **negative value** attenuates the BT signal sent to the AUX bus.
- **0 dB** applies unity gain.
- A **positive value** increases the AUX send level.

These controls create an **independent AUX mix without changing the main BT channel-fader level**. The exact range and adjustment step should be taken from the values displayed by the installed firmware.

BUTTON LIGHT:

The **Button Light** parameter selects the RGB colour assigned to the BT source's **ON/START button**.

The selected colour helps identify the Bluetooth channel and distinguish it from microphone, line, telephone or digital sources. The button can use different brightness or colour states to indicate whether the channel is **off**, **in standby** or **active**. The available selection normally consists of the console's preset or user-configured RGB colours.

FADER BAR LIGHT:

The **Fader Bar Light** parameter selects the RGB colour of the LED bar associated with the BT source.

The LED bar provides visual information about:

- the **fader position or attenuation level**;
- the **source type**;
- the colour assigned to the BT input.

The selected colour does not affect audio processing. Its purpose is to provide **immediate visual identification of the Bluetooth source** on the console surface.

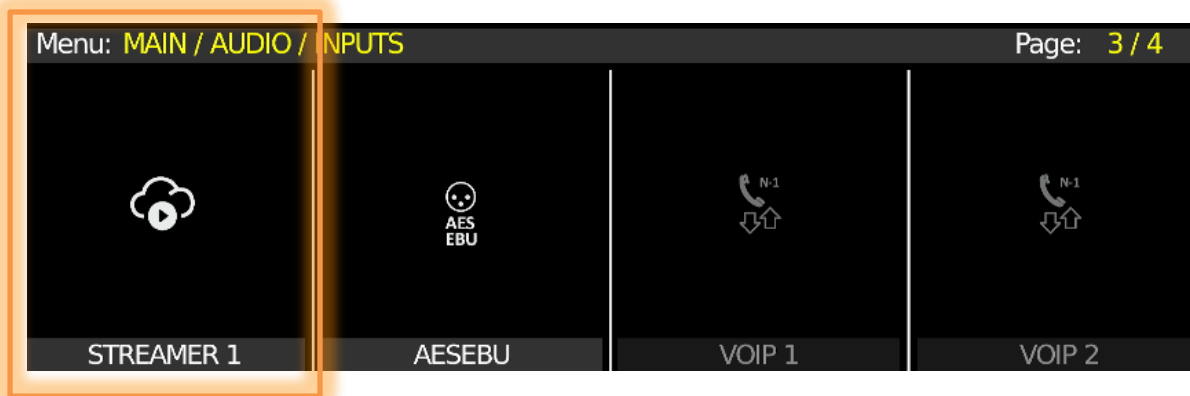
STREAMING

The **STREAMING input** is the logical audio source generated by the network decoder integrated into the **AxelTech Oxygen 3000 Plus**. It receives a mono or stereo audio stream through the console's Ethernet connection, decodes it and makes it available within the mixer as a normal input source.

The STREAMING input must not be confused with the console's streaming output:

- **Streaming Input / Decoder:** receives and decodes audio coming from the network.
- **Streaming Output / Encoder:** encodes a selected mixer signal and transmits it over the network.

The Oxygen 3000 Plus provides **one Streaming input and one Streaming output**, independently of the number of Audio I/O Boards installed. Typical applications include **remote audio contributions, studio-to-transmitter links, studio-to-studio links, web-radio feeds, backup programme sources and network audio distribution**.



Connecting a streaming source

The audio stream enters the console through its **Ethernet/LAN connection** rather than through an analogue, USB or AES/EBU audio connector.

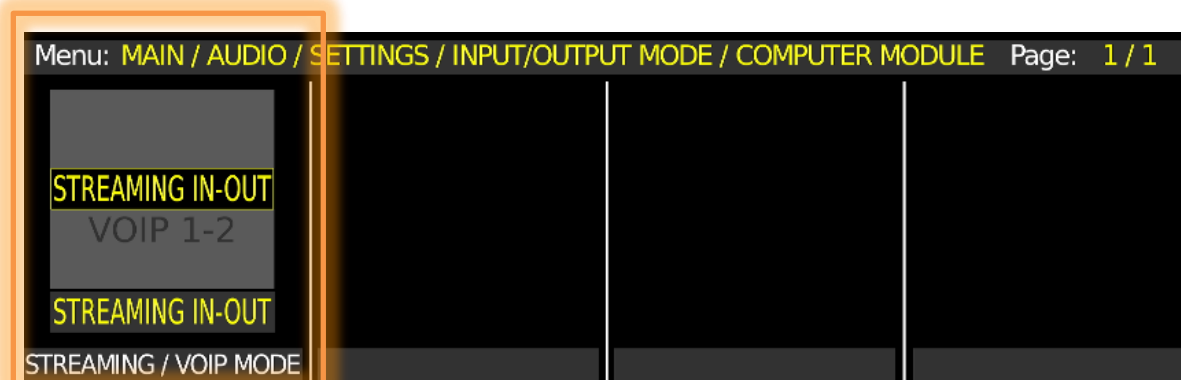
The general connection procedure is:

1. Connect the console's **LAN port** to the local network using an Ethernet cable.
2. Verify the console's IP configuration under **GENERAL SETTINGS → COMMUNICATIONS → TCP-IP**. The address can be assigned manually or provided by a DHCP server, according to the network configuration.
3. From a computer connected to the same network, enter the console's IP address in a compatible web browser to open the **Oxygen web interface**.
4. Open the section dedicated to the **Streaming Decoder/Input**.
5. Configure the parameters required by the source. Depending on the selected protocol, these may include the **server or source address, stream URL, destination or listening port, multicast group and decoding format**.
6. Apply the configuration and start or enable the decoder. Verify that the interface indicates a valid connection and that audio is present on the input meter.
7. On the mixer surface, rotate or press the encoder of the required channel and select **STREAMING** from the source list.
8. Route the channel to the required bus, such as **PGM, SUB, AUX 1 or AUX 2**, and open it according to its ON MODULE configuration.

For a server-based stream, such as an Icecast service, the decoder normally connects to the configured **stream address or URL**. For a direct RTP or UDP feed, the console must use the correct **IP address, port and unicast or multicast configuration** provided by the transmitting device.

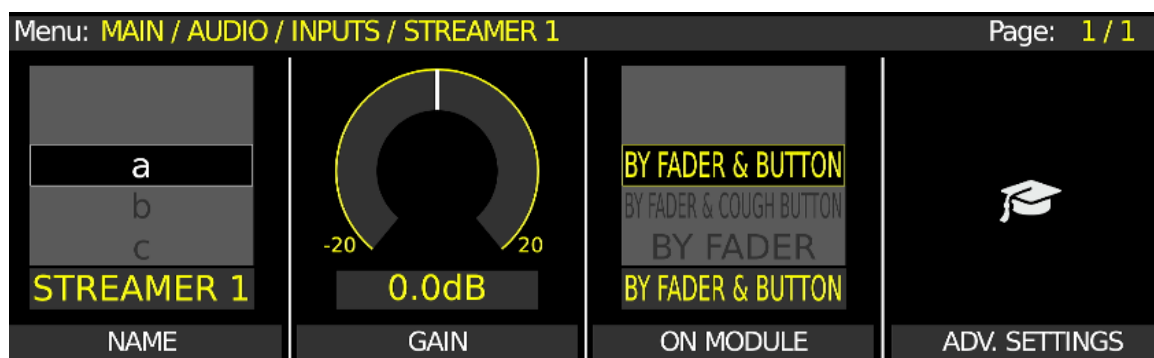
The Oxygen 3000 Plus supports **IPv4, IPv6, TCP/IP, UDP, RTP and RTSP**, with both **unicast and multicast** operation. Both **CBR and VBR streams** are supported for input decoding. AxelTech also specifies compatibility with Icecast streaming using **MP3, Opus and Ogg Vorbis**

ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

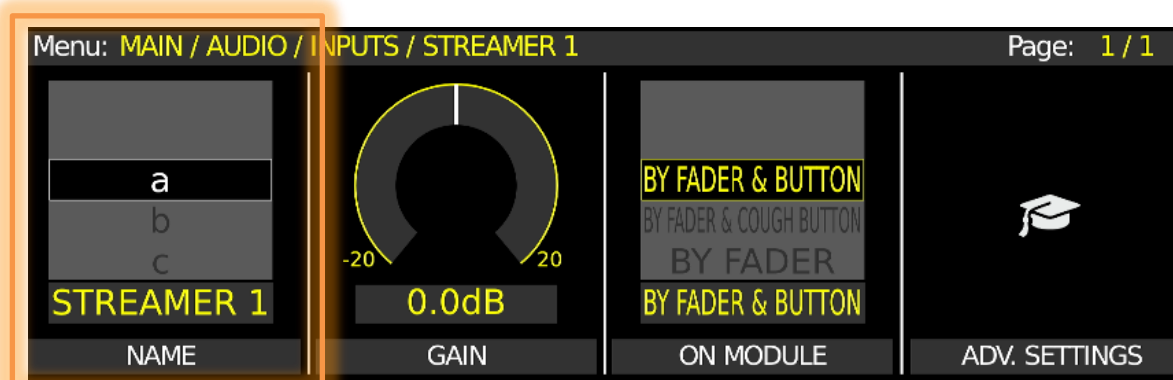


INPUTS SETTING (STREAMING)

Here we can find the source parameters (STREAMING). In this page you can modify/set different parameters of the **STREAMING**. The main modality are: **NAME, GAIN, ON MODULE** and **ADV SETTINGS**.



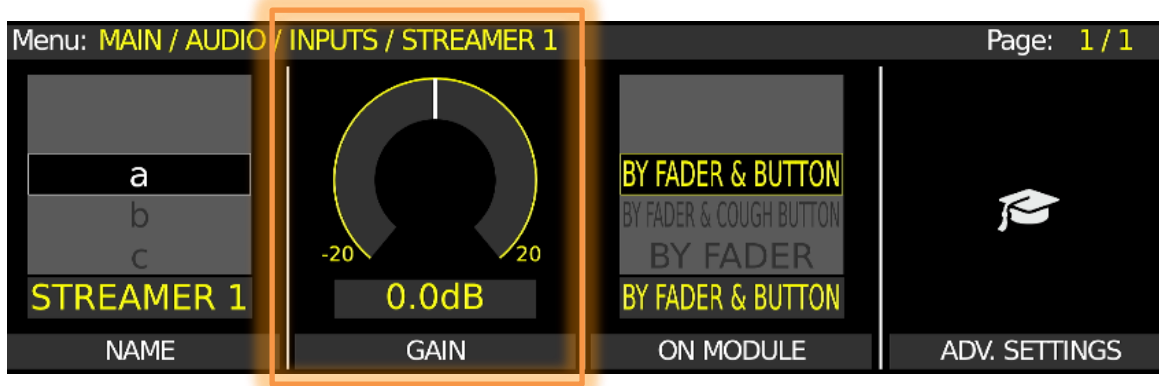
NAME:



The **Name** parameter assigns a customised label to the **STREAMING** source. The selected name is displayed whenever the source is assigned to a physical or virtual mixer channel.

The label can contain up to **8 characters**. Typical names include **STREAM**, **REMOTE**, **STL**, **WEBRADIO** or **BACKUP**. Changing this parameter affects only the source label shown by the console; it does not modify the stream name, URL or network configuration.

GAIN:

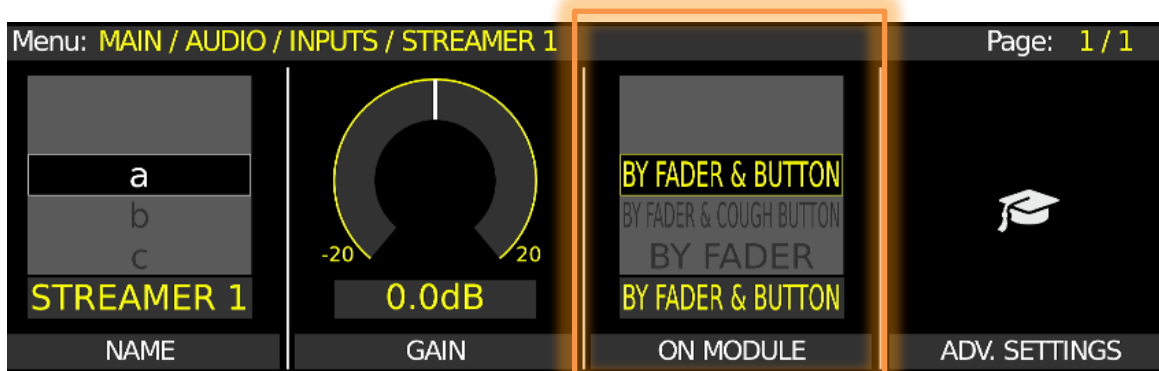


The **Gain** parameter adjusts the decoded stream level before it is processed by the mixer channel and routed to the selected buses. **Range: -20.0 dB to 20.0 dB. Adjustment step: 0.1 dB.**

A negative value attenuates a stream that is too loud, while a positive value increases a low-level contribution. Gain should be adjusted while observing the **pre-fader level meter**, ensuring that the decoded audio does not reach the clipping point.

The Gain parameter affects only the audio level inside the mixer. It does not change the stream's original encoder bitrate, transmission level or network configuration.

ON MODULE:



The **On Module** parameter determines how the STREAMING source is opened, closed or temporarily muted from the mixer channel to which it is assigned.

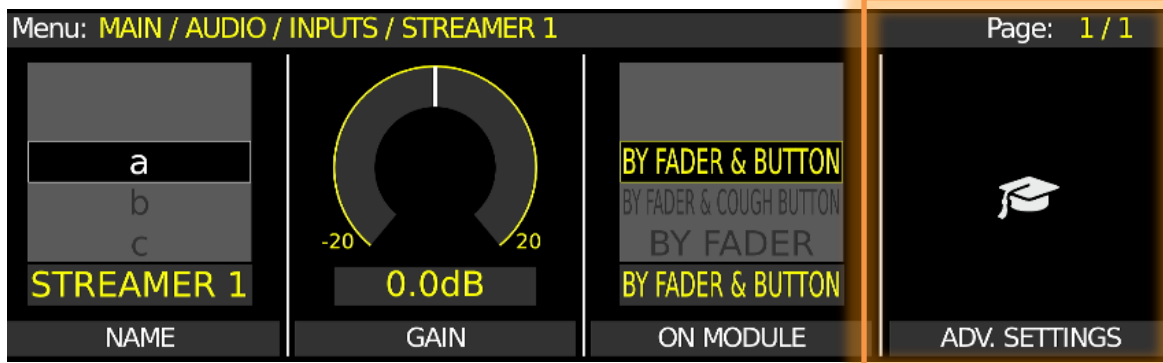
The selectable options are:

- **FADER & BUTTON:** the channel is enabled through the combined operation of the **ON/START button and fader**. The button enables the channel, while the fader controls its opening and level.
- **BY FADER:** raising the fader above the configured activation threshold opens the channel. Lowering it below the threshold closes the channel.
- **BY BUTTON:** the channel is opened or closed through the **ON/START button**. The fader controls the audio level without determining the channel's ON/OFF status.
- **BY FADER & COUGH BUTTON:** the fader controls normal channel activation, while a configured **COUGH button temporarily mutes the STREAMING source**. Releasing the button restores the audio, provided that the fader remains above the activation threshold.

- **ALWAYS ON:** the STREAMING source remains enabled independently of the ON/START button. The fader can still control the signal sent to post-fader buses.
- **ALWAYS OFF:** the STREAMING source remains disabled and cannot be opened using the channel controls.

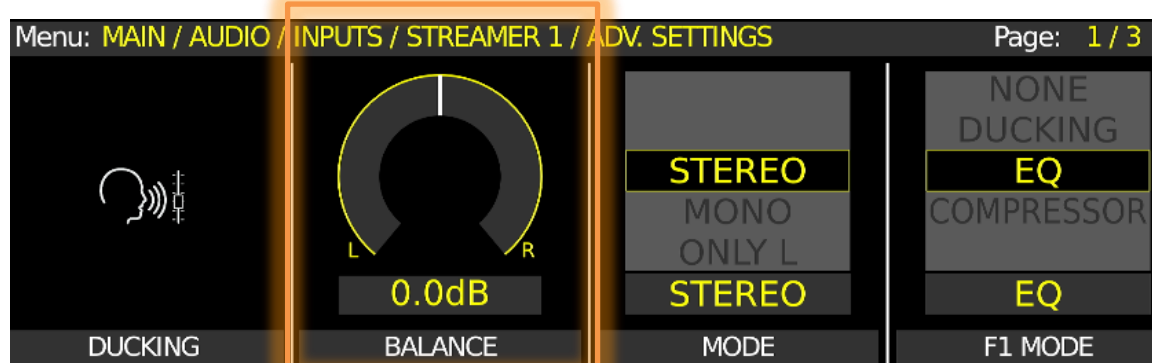
Selecting **ALWAYS ON** does not establish the network connection automatically. It only keeps the mixer channel enabled; audio will be available only when the decoder is correctly configured and receiving a valid stream.

ADV. SETTINGS:



The **Advanced Settings** menu opens the additional processing, routing, programmable-button and visual-identification parameters associated with the STREAMING input.

DUCKING:

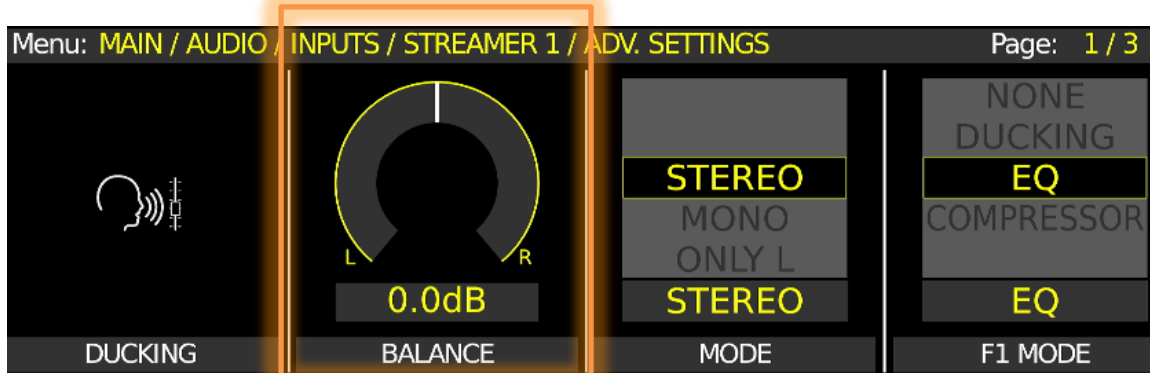


The **Ducking** function allows the STREAMING source to participate in the console's automatic attenuation system. It is particularly useful when the decoded stream contains background music that must be reduced automatically while a presenter is speaking. The two operating roles are:

- **SLAVE MODE:** determines whether the STREAMING input is attenuated when an assigned master source becomes active.
 - **ON:** the STREAMING source operates as a ducking slave.
 - **OFF:** automatic attenuation is disabled for the STREAMING source.

Slave Mode enables **attenuation on the STREAMING input**.

BALANCE:



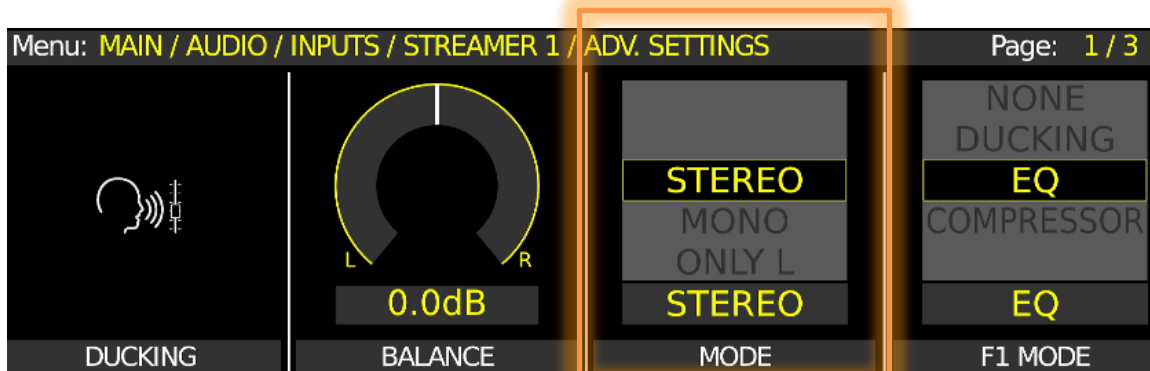
The **Balance** parameter adjusts the relative level of the Left and Right channels of a decoded stereo stream. The control positions are:

- **LEFT:** progressively attenuates the Right channel.
- **CENTER:** maintains equal level on the Left and Right channels.
- **RIGHT:** progressively attenuates the Left channel.

The centre position should normally be used to preserve the original stereo image. Balance can be adjusted to compensate for an incorrectly balanced stream or for a difference between the two received channels.

When the decoder is receiving a mono stream, Balance normally has no practical benefit unless the mono signal is present on only one side of the decoded stereo path.

MODE:



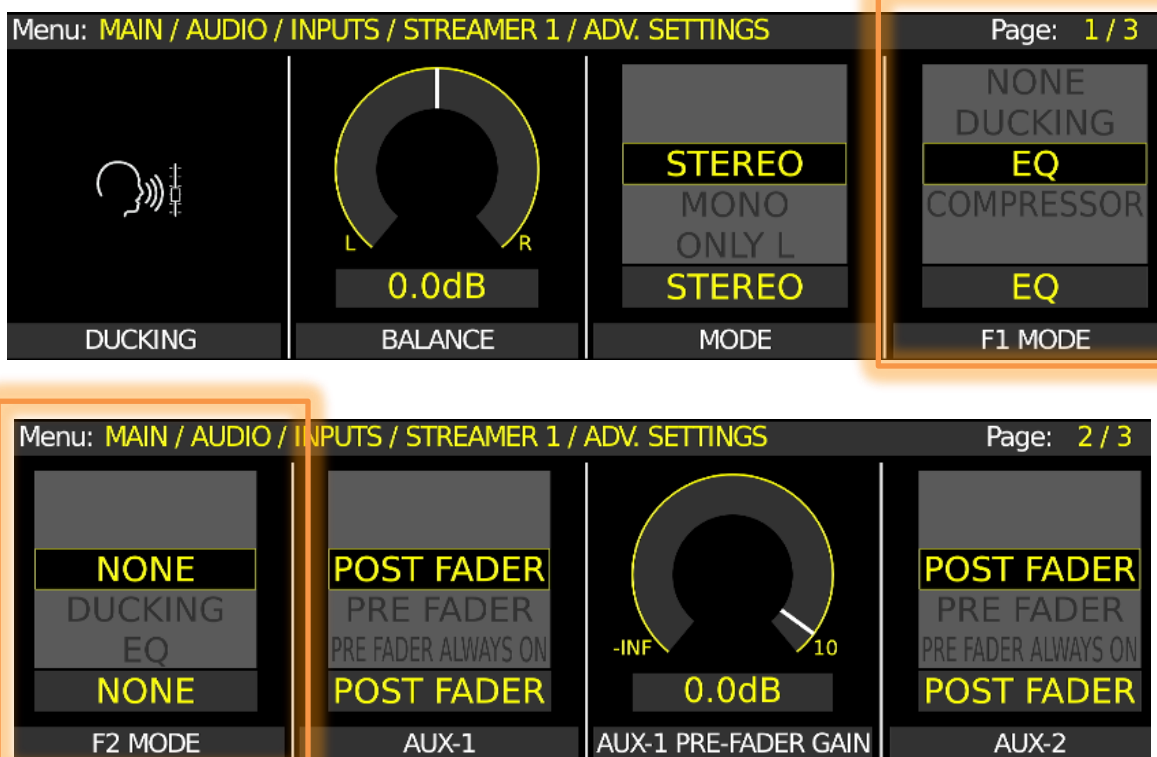
The **Mode** parameter determines how the Left and Right channels of the decoded STREAMING signal are processed before being routed to the mixer buses. The available options are:

- **STEREO:** preserves the original Left and Right channels of the decoded stream. This is the normal setting for stereo music and programme feeds.
- **MONO:** combines the Left and Right channels into a single centred mono signal. This mode is useful when stereo separation is unnecessary or when the stream must be reproduced consistently through both output channels.
- **ONLY L:** retains only the **Left channel** of the decoded stream and excludes the Right channel. It is useful when the required contribution is carried exclusively on the Left side.

- **ONLY R:** retains only the **Right channel** of the decoded stream and excludes the Left channel. It is useful when the required contribution is carried exclusively on the Right side.
- **INV. L:** reverses the polarity of the **Left channel by 180°**, while leaving the Right channel unchanged.
- **INV. R:** reverses the polarity of the **Right channel by 180°**, while leaving the Left channel unchanged.
- **INV. L&R:** reverses the polarity of **both the Left and Right channels by 180°**.

When the external stream is natively mono, the decoder configuration should first be checked in the **Oxygen Web Interface**. The Mode parameter acts on the decoded audio inside the mixer and does not change the network stream format.

F1 MODE / F2 MODE:



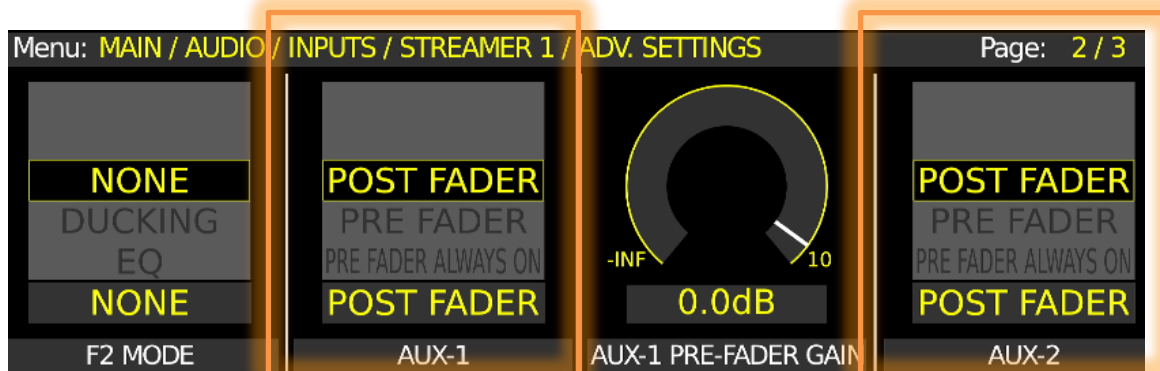
The **F1 Mode** and **F2 Mode** parameters assign a function to the two programmable buttons associated with:

- **NONE:** no function is assigned. Pressing the corresponding F button has no effect on the STREAMING source.
- **DUCKING:** assigns the ducking control or status indication to the F button. Its behaviour depends on the STREAMING source's Ducking configuration: If STREAMING is configured as a **slave source**, its attenuation is controlled automatically by the associated master source.
- **EQ:** enables or disables the equalizer associated with the STREAMING source.
 - If the EQ Mode is **UNLOCKED**, the F button can switch the equalizer on or off.
 - If it is **LOCKED ON**, the equalizer remains permanently enabled.
 - If it is **LOCKED OFF**, the equalizer remains permanently disabled.

-COMPRESSOR: enables or disables the compressor associated with the STREAMING source. When enabled, the compressor controls excessive level variations according to its configured **Threshold, Ratio, Attack, Release and Makeup Gain** parameters. If the compressor status is locked, the F button cannot override the selected state.

When the STREAMING source is configured only as a ducking slave, the attenuation is controlled automatically by the assigned master source rather than manually by the F button.

AUX 1 / AUX 2:

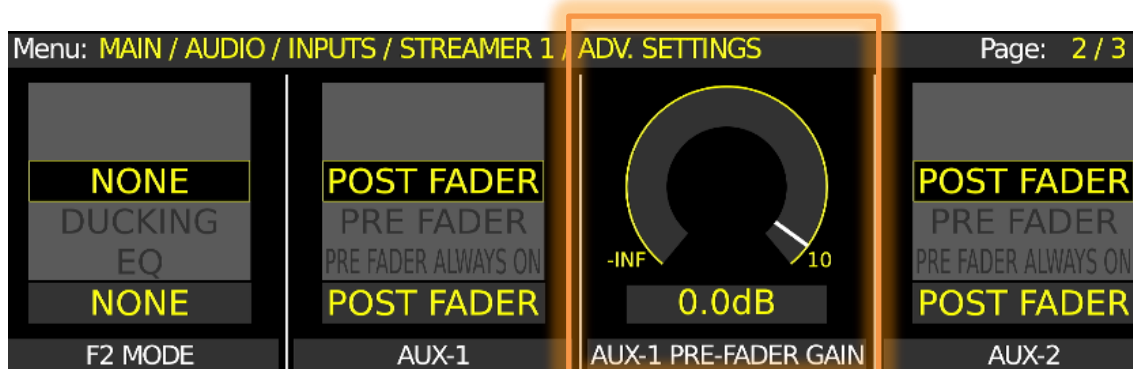


The **AUX 1** and **AUX 2** parameters determine where the STREAMING signal is taken in relation to the channel fader and ON/START status before being sent to the corresponding auxiliary bus. The selectable modes are:

- **POST-FADER:** the AUX signal follows the channel fader and ON/START status. Any change made with the fader is reflected in the auxiliary output.
- **PRE-FADER:** the AUX signal is independent of the channel fader but remains controlled by the channel's ON/OFF status.
- **PRE-FADER ALWAYS ON:** the AUX signal is independent of both the channel fader and ON/START button. The decoded stream remains available on the auxiliary bus whenever the network decoder is receiving valid audio.

POST-FADER is suitable when the AUX level must follow the main mix. **PRE-FADER** is useful for independent monitoring or recording, while **PRE-FADER ALWAYS ON** is suitable for permanently available service or distribution feeds.

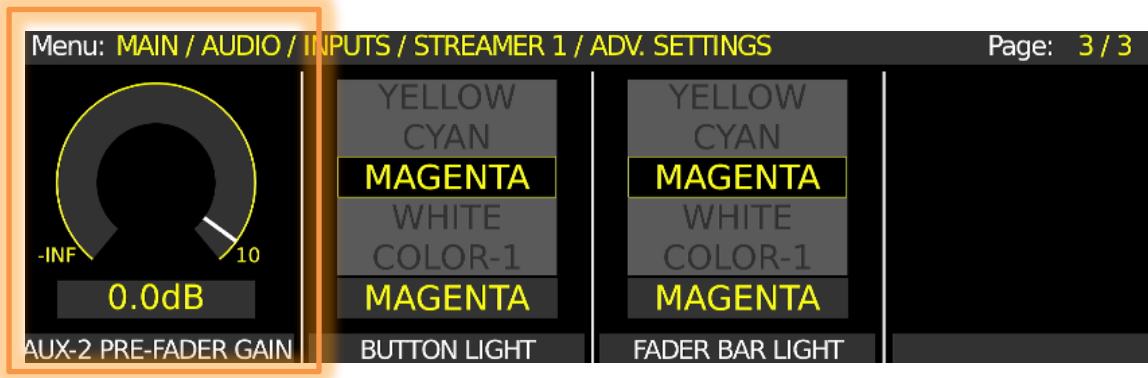
AUX 1 PRE-FADER GAIN / AUX 2 PRE-FADER GAIN:



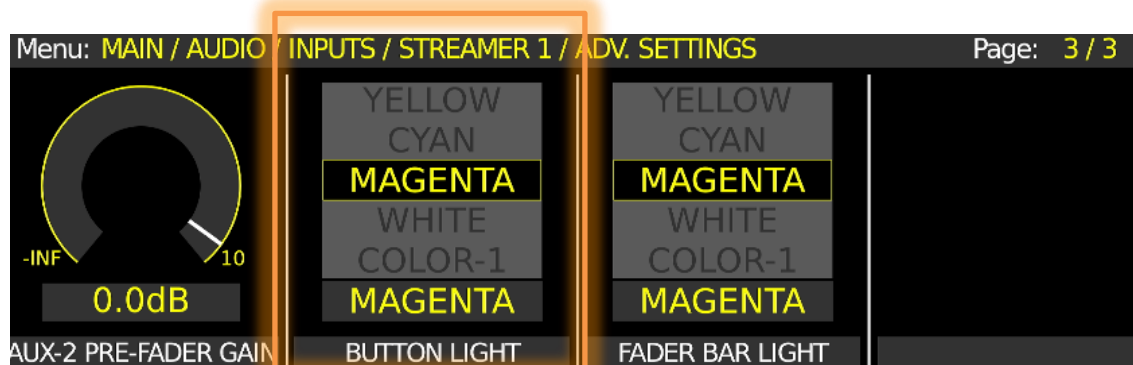
These parameters adjust the STREAMING level sent to **AUX 1** or **AUX 2** when the corresponding auxiliary bus is operating in a pre-fader mode.

- A **negative value** attenuates the decoded stream sent to the AUX bus.
- **0 dB** applies unity gain.
- A **positive value** increases the pre-fader AUX level.

This adjustment creates an **independent auxiliary mix without changing the main STREAMING channel level**.



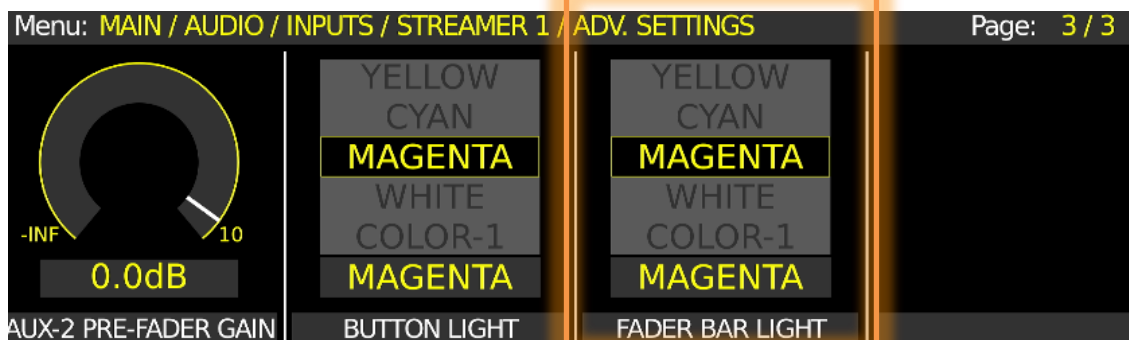
BUTTON LIGHT:



The **Button Light** parameter selects the RGB colour assigned to the STREAMING source's **ON/START button**. The selected colour helps the operator distinguish the network stream from microphone, telephone, Bluetooth, USB and analogue sources. According to the channel state, the button can indicate that the source is:

- **OFF:** the channel is disabled;
- **STANDBY/WARM:** the channel is enabled but the fader is closed;
- **ON:** the channel is open.

FADER BAR LIGHT:

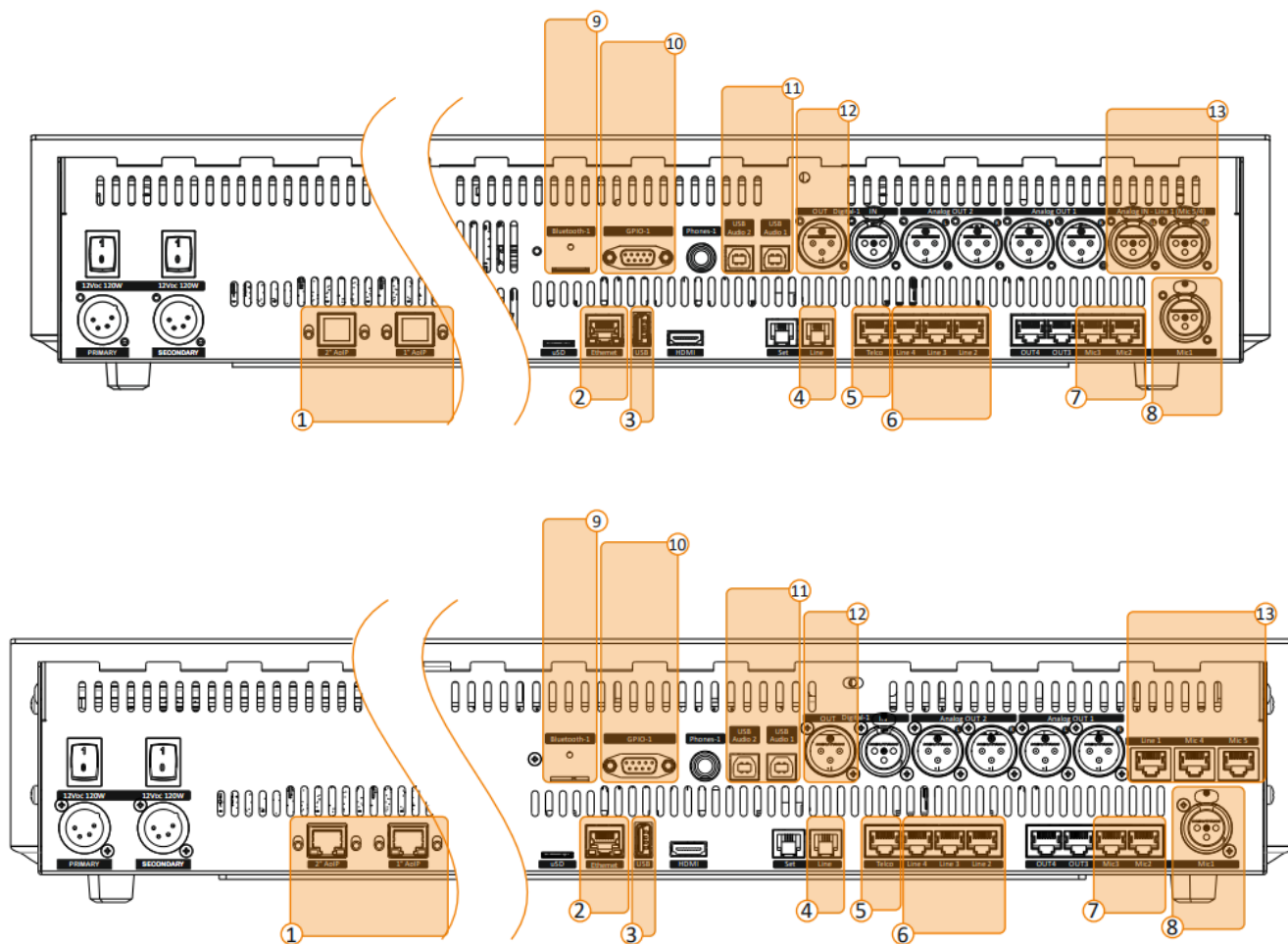


The **Fader Bar Light** parameter selects the RGB colour of the LED bar associated with the STREAMING source. The LED bar provides visual information about:

- the **fader position or attenuation level**;
- the **source category**;
- the colour assigned to the STREAMING input.

This setting affects only the visual identification of the source and does not modify the decoded audio, stream connection or channel routing.

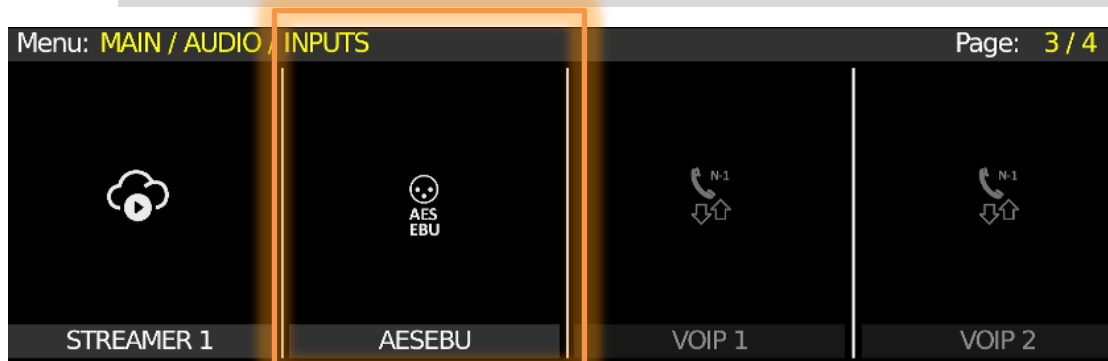
DIGITAL AES/EBU



Important: The audio input connections for DIGITAL AES/EBU are located on the first Audio Board, at position 12.

The **AES/EBU source** is a professional digital audio input integrated into the **Oxygen 3000 Plus Digital Mixing Console**. It uses the AES3 digital audio standard to receive two audio channels—normally configured as a stereo Left/Right signal—through a **single female XLR connector with a balanced 110 Ω connection**. This input can be used to connect professional digital equipment such as **audio processors, playout systems, digital players, codecs or other mixers** equipped with an AES/EBU output. Since the signal remains in the digital domain, unnecessary digital-to-analogue and analogue-to-digital conversions are avoided, helping to preserve **audio quality, low noise and signal integrity**. The built-in **Sample Rate Converter** allows the console to accept incoming sampling frequencies from **32 to 192 kHz** and adapt them to its internal processing rate. An appropriate **110 Ω AES/EBU cable** should be used to ensure reliable data transmission and prevent synchronization errors.

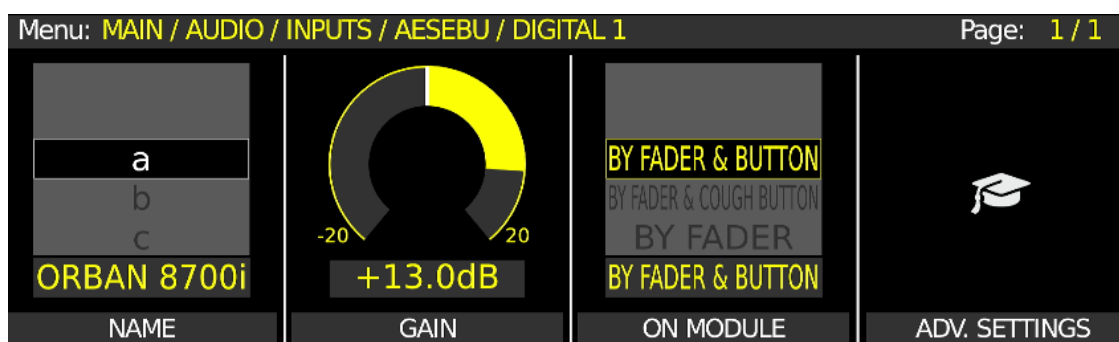
ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.



ATTENTION: For the selected option to become available, verify that it has been configured as an audio input for the mixer. For further information, refer to the **INPUT SWITCH** description provided at the beginning of the **SETTINGS** section.

INPUTS SETTING (AESEBU)

Here we can find the source parameters (AESEBU). In this page you can modify/set different parameters of the AESEBU. The main modality are: **NAME**, **GAIN**, **ON MODULE** and **ADV SETTINGS**.



ATTENTION: The main settings available in this interface, including those found under ADV. SETTINGS, are identical to the settings provided for the **STEREO** input.

VOIP

The **VOIP** section configures the logical input sources associated with the OXYGEN 3000 Plus Computer Module when it is operating in VOIP mode.

It can normally be accessed through: **MAIN → AUDIO → INPUTS → VOIP → VOIP 1**

or, for the second available path: **MAIN → AUDIO → INPUTS → VOIP → VOIP 2**

Depending on firmware, VOIP may instead be grouped under PHONE:

MAIN → AUDIO → INPUTS → PHONE → VOIP → VOIP 1

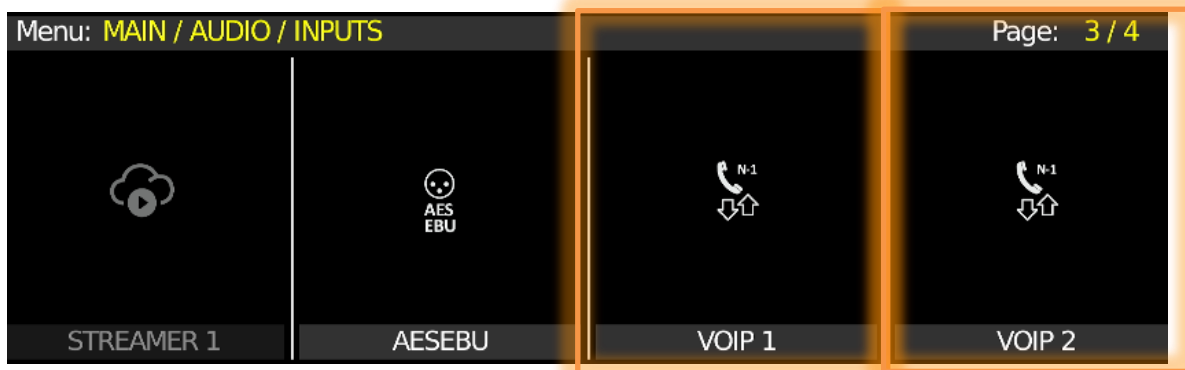
Before the VOIP sources can be used, the Computer Module must normally be configured through:

MAIN → AUDIO → SETTINGS → INPUT/OUTPUT MODE → COMPUTER MODULE → STREAMING/VOIP MODE → VOIP

When VOIP mode is selected, the Computer Module can provide one or two logical communication paths:

- **VOIP 1**
- **VOIP 2**

The OXYGEN 3000 Plus supports VoIP sources as part of its telephone and Talk Show integration. The current system can manage different telephone-line types, including POTS, BT, TELCO and VoIP.



The VOIP page includes:

- **NAME**
- **GAIN**
- **ON MODULE**
- **ADV. SETTINGS**

Important: VOIP configures the source inside the console. It does not configure the communication application, SIP account, server, password, audio-device selection or remote destination. These parameters must be configured separately in the associated software.

VOIP signal structure:

A VOIP connection contains two separate audio directions.

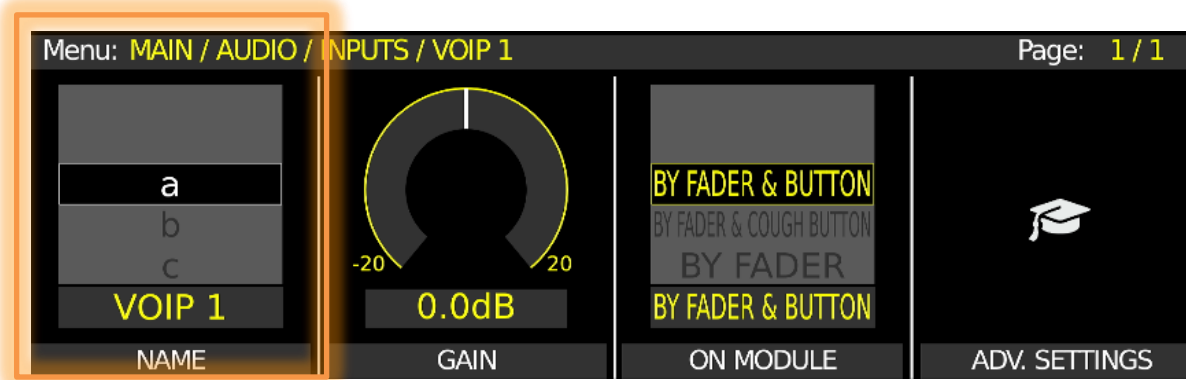
- **Receive path — remote participant to console:** The receive path carries the remote participant's audio from the computer application into the OXYGEN 3000 Plus.
- **Transmit path — console to remote participant:** The transmit path carries audio from the console to the remote participant. The return should normally use an **N-1** or clean-feed signal. This allows

the participant to hear the required programme sources without receiving their own voice back with network delay.

Warning: if the VOIP receive signal is included in its own transmit return, the remote participant may hear echo or delayed self-monitoring. Verify the N-1 configuration before putting the call On Air.

INPUTS SETTING (VOIP)

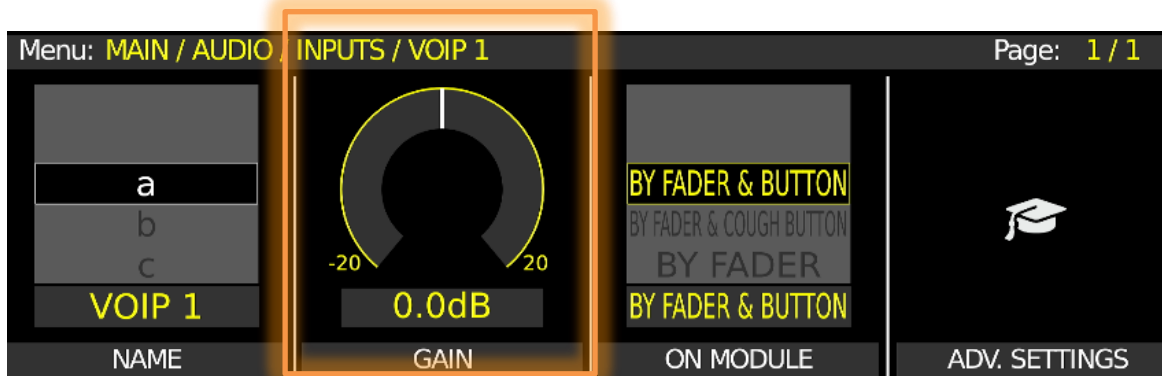
NAME:



NAME assigns a customised label to the selected VOIP source. The label normally contains up to **8 characters**, following the standard input-source naming structure used by the OXYGEN 3000 Plus. The input names can be edited from the console and are used to identify sources throughout the routing system. A typical editing name procedure is:

1. Select **NAME**.
2. Rotate the associated encoder to select a character.
3. Press the encoder to confirm the character.
4. Repeat the operation for the remaining positions.
5. Press **ESC** when the name is complete.

Important: changing NAME modifies only the label displayed by the console. It does not rename the audio device inside the computer, change the communication account or select a different application.

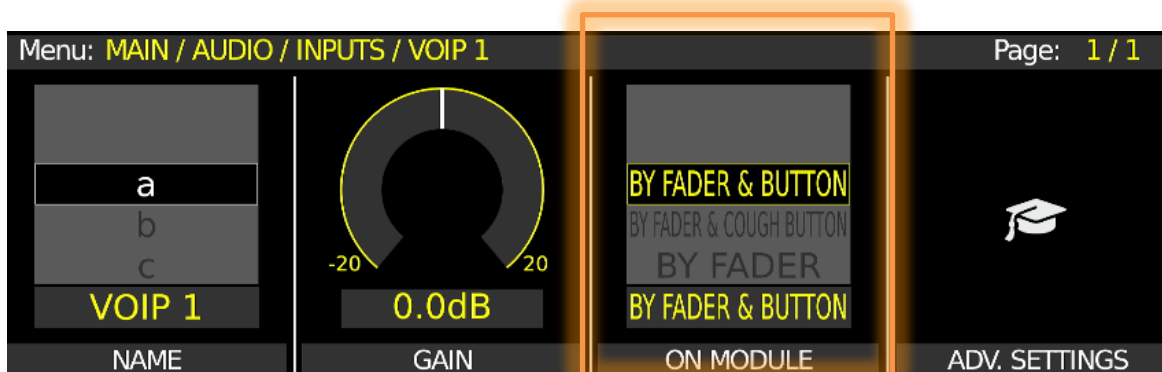
GAIN:

GAIN adjusts the level of the VOIP audio received by the OXYGEN 3000 Plus. **Range: -20.0 dB to +20.0 dB.**

Adjustment step: 0.1 dB.

- Negative values attenuate the incoming VOIP signal.
- **0.0 dB** applies unity gain. The incoming VOIP signal is passed without additional digital attenuation or amplification.
- Positive values amplify the incoming VOIP signal.

Warning: excessive positive GAIN can increase background noise, codec artefacts and computer-generated noise. It may also overload the compressor, AUX buses or programme path.

ON MODULE:

ON MODULE determines how the channel carrying the VOIP source changes between its logical OFF and ON states. The selectable options are:

- **FADER & BUTTON:** The VOIP channel is activated through the combined operation of: The **physical fader** - The **ON/START button**. The source becomes active when the required button state is enabled and the fader is raised above the configured threshold. **Important:** the VOIP call may remain connected while the channel is OFF. FADER & BUTTON controls only its routing through the console.
- **BY FADER:** The logical channel state follows the physical fader. BY FADER is useful when fast operation is required and opening or closing the participant must follow the fader directly.n

Warning: raising the fader may immediately place the remote participant On Air when PGM or another programme bus is already assigned.

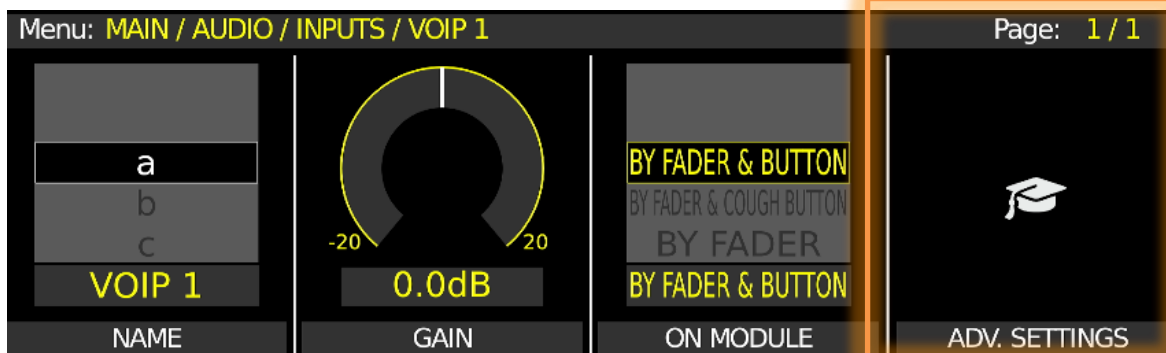
- **BY BUTTON:** The logical VOIP channel state is controlled by the ON/START button. **Important:** closing the fader does not necessarily switch the channel OFF. Always check the ON/START illumination.
- **ALWAYS ON:** The VOIP source remains logically enabled. **Warning:** ALWAYS ON can leave the VOIP source active in pre-fader, direct or auxiliary paths even when the main fader is closed.
- **ALWAYS OFF:** The VOIP source remains logically disabled. The configuration is retained, but the channel cannot normally be activated from the standard surface controls. **Important:** ALWAYS OFF does not delete the NAME, GAIN or ADV. SETTINGS configuration. It also does not necessarily disconnect an active call inside the computer application.

The OXYGEN 3000 Plus uses these operating modes to define how a source responds to the physical fader and ON/START button.

Important: ON MODULE controls the VOIP source inside the console. It does not necessarily answer, connect or terminate the software call. Call control is handled separately by the communication application or by an F button assigned to VOIP MODULE.

ADV. SETTINGS:

ADV. SETTINGS opens the additional processing, routing and surface-control parameters associated with the VOIP source.



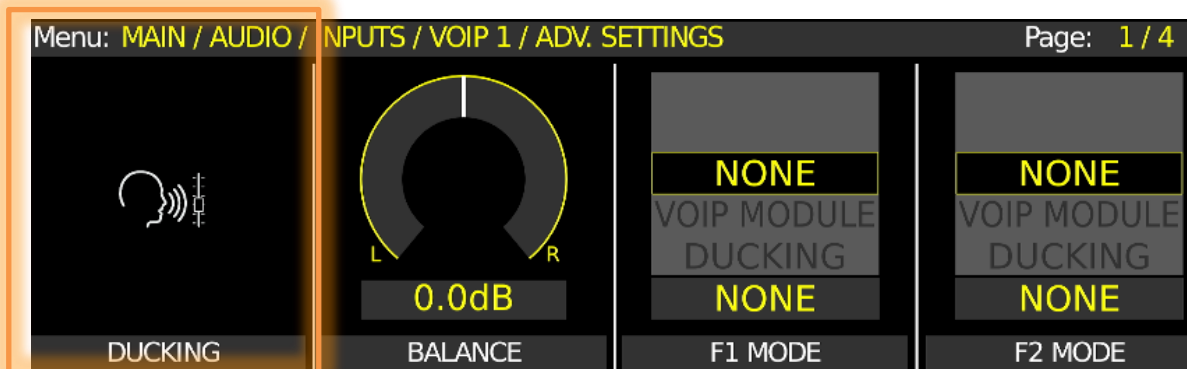
Depending on firmware, this section may include:

- DUCKING
- DYN. COMP.
- MR LIGHT RGB
- BALANCE
- F1 MODE
- F2 MODE
- AUX-1
- AUX-1 PRE-FADER GAIN
- AUX-2
- AUX-2 PRE-FADER GAIN
- BUTTON LIGHT

- **FADER BAR LIGHT**

Important: ADV. SETTINGS mainly processes and routes the VOIP receive signal. The return sent to the remote participant remains a separate path and must be configured through the Computer Module, communication application or corresponding output assignment.

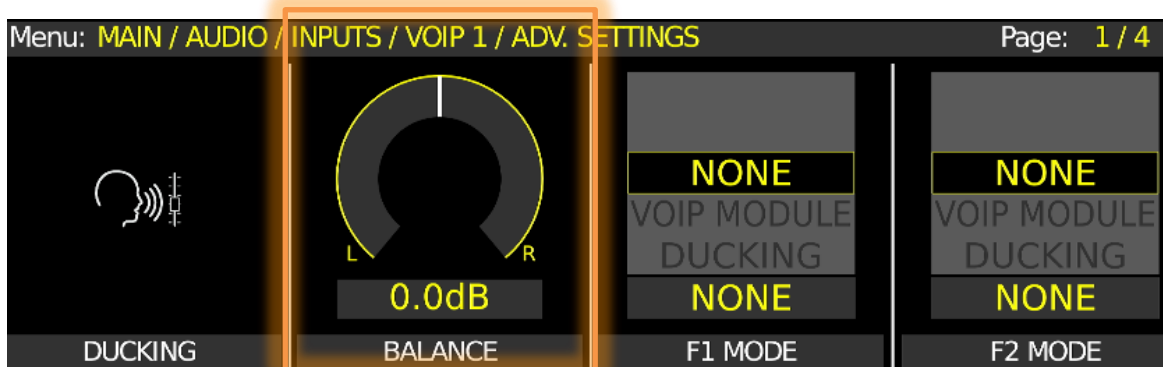
DUCKING:



DUCKING determines how VOIP interacts with the automatic attenuation system. The VOIP source can operate as the **SLAVE mode**, being attenuated when another master becomes active.

ATTENTION: A detailed description of this mode can be found in the section dedicated to the MIC input.

BALANCE:



BALANCE adjusts the relative level of the Left and Right channels of the VOIP receive source.

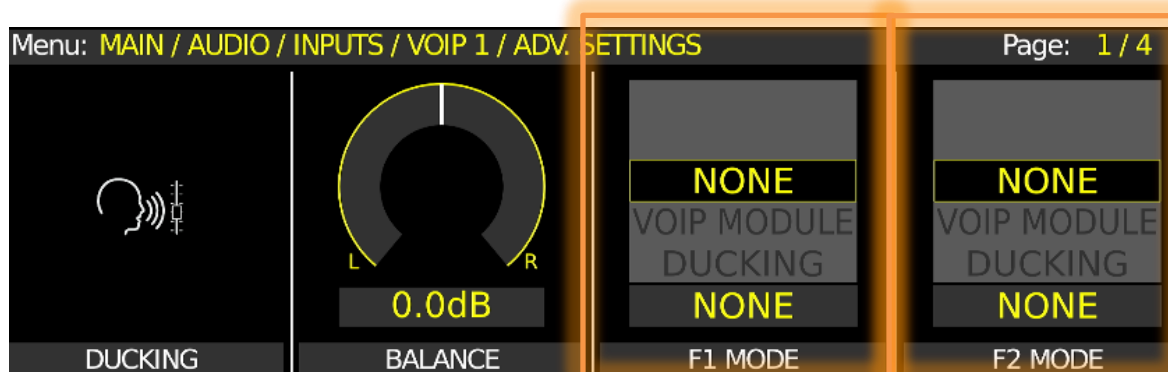
The available adjustment extends from:

- Full **LEFT** : Reduces or removes the Right-channel contribution, moving the VOIP source towards the Left side.
- Through **CENTRE**: Applies equal level to the Left and Right channels. This is the normal setting when the communication application supplies a correctly centred signal.
- To full **RIGHT**: Reduces or removes the Left-channel contribution, moving the VOIP source towards the Right side.

The adjustment normally uses **1.0 dB steps**, following the console's BAL/PAN control structure.

Important: BALANCE affects only the VOIP receive source inside stereo buses. It does not change the return heard by the remote participant.

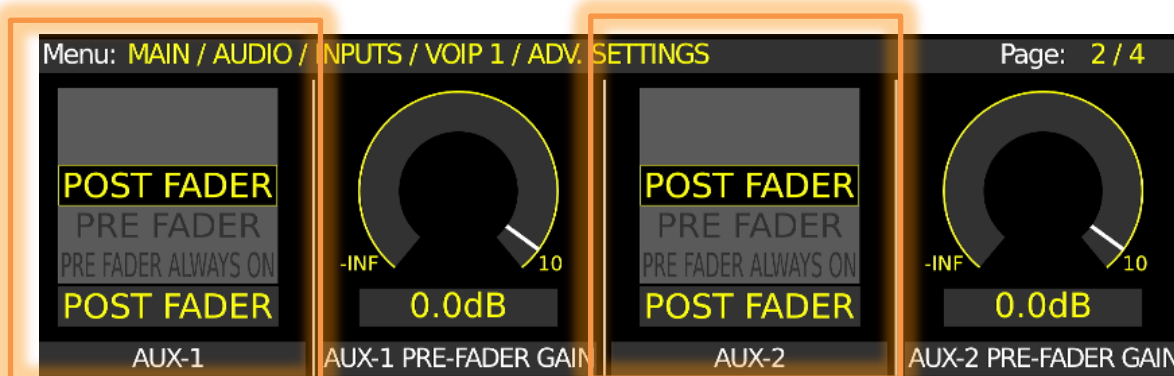
F1 MODE / F2 MODE:



F1 MODE and **F2 MODE** independently assign functions to the two programmable F buttons of the channel carrying VOIP. Depending on firmware, the selectable options may include:

- **NONE:** The selected F button performs no action.
- **VOIP MODULE:** Sends a control command to the associated VOIP software module. The exact action depends on the installed Computer Module and communication application. **Important:** VOIP MODULE is application-dependent. Verify the command assigned by the installed software before using it during a live programme.
- **DUCKING:** Activates or deactivates ducking when DUCKING STATUS is set to **UNLOCKED**.

AUX-1 / AUX-2:

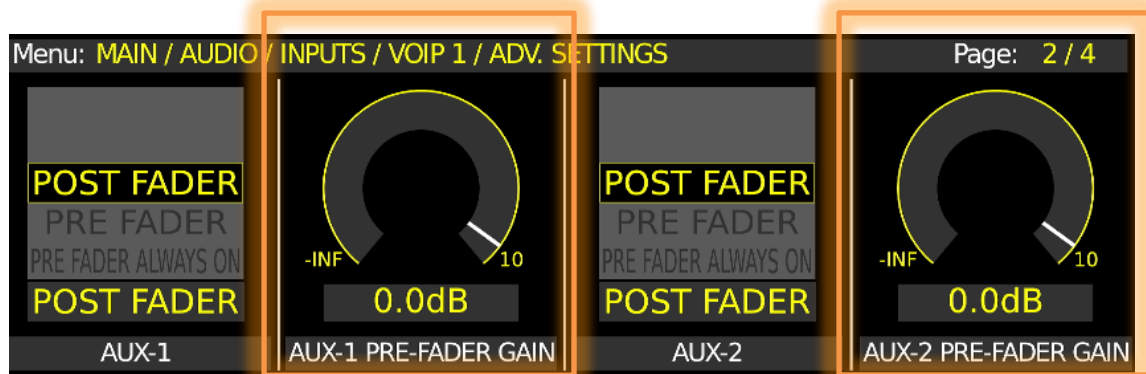


AUX-1 and **AUX-2** independently determine how the VOIP receive source is sent to the corresponding auxiliary bus. The selectable options are:

- **POST FADER:** The VOIP level in the AUX bus follows the physical channel fader. POST FADER is suitable when the AUX destination must follow the On Air VOIP level.
- **PRE-FADER:** The VOIP signal is taken before the physical fader.
- **PRE-FADER ALWAYS ON:** The VOIP signal remains present independently from: The **physical fader** - The **ON/START state**. This mode may be used for a permanent monitoring or communication path. **Warning:** PRE-FADER ALWAYS ON can leave a private VOIP conversation audible on an AUX output while the main channel appears to be OFF. Verify privacy and destination routing carefully.

The OXYGEN 3000 Plus supports post-fader, pre-fader and always-on pre-fader AUX operation.

AUX-1 PRE-FADER GAIN / AUX-2 PRE-FADER GAIN:



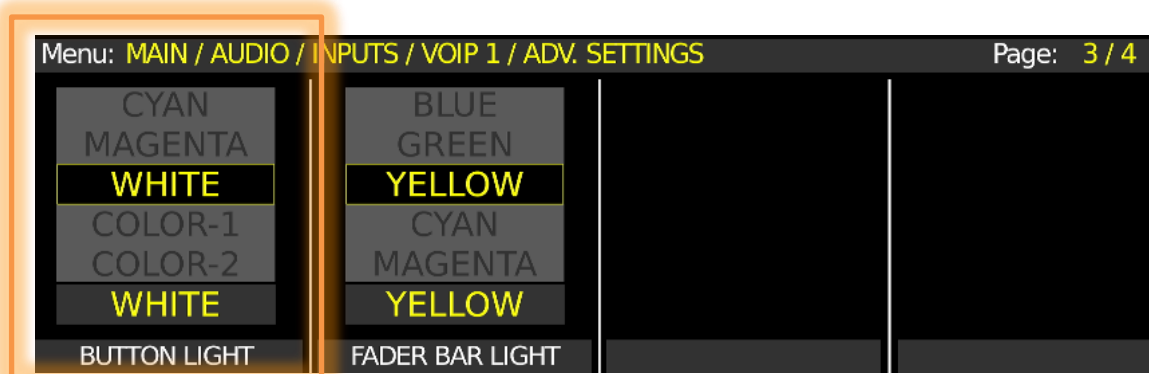
These parameters independently set the VOIP receive level sent to the AUX-1 and AUX-2 pre-fader paths.

Range: -INF to +10.0 dB. **Adjustment step:** 1.0 dB.

- Negative values attenuate VOIP in the selected AUX bus.
- 0.0 dB Apply unity gain to the pre-fader path.
- Positive values increase the VOIP level in the selected AUX bus.

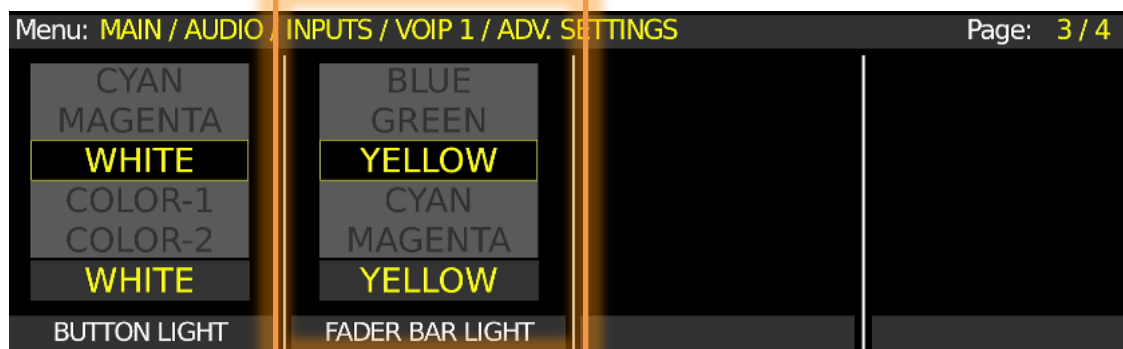
Warning: excessive positive gain may overload the AUX bus even when the main VOIP channel level is correct.

BUTTON LIGHT:



BUTTON LIGHT selects the colour preset used by the illuminated buttons of the channel carrying VOIP.

Important: BUTTON LIGHT changes the colour assignment only. It does not change the function, source routing or meaning of flashing and steady button states.

FADER BAR LIGHT:

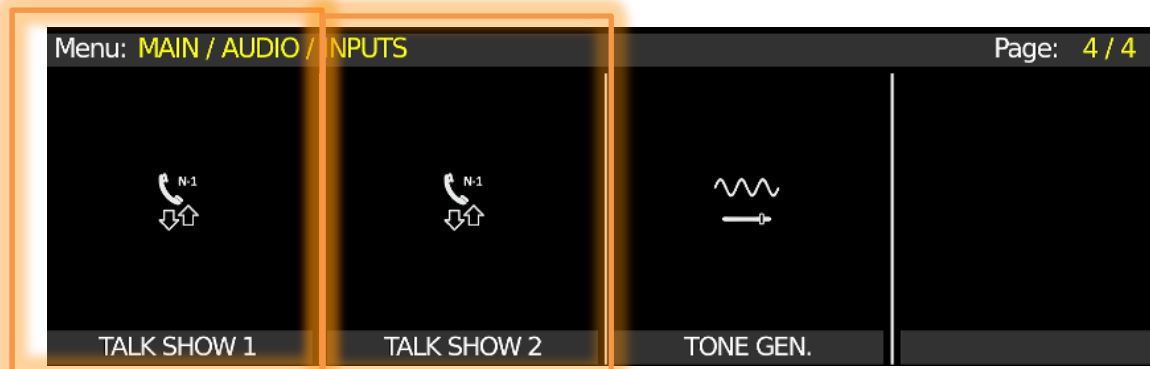
FADER BAR LIGHT selects the colour preset used by the LED bar associated with the VOIP channel fader.

TALK SHOW

The **TALK SHOW** section configures the surface-control, auxiliary-routing and illumination parameters associated with the OXYGEN 3000 Plus Talk Show System sources.

It can normally be accessed through: **MAIN → AUDIO → INPUTS → TALK SHOW → TALK SHOW 1**

or, for the second logical source: **MAIN → AUDIO → INPUTS → TALK SHOW → TALK SHOW 2**



The Talk Show System is an optional software function designed to simplify the management of multiple telephone calls. It can manage up to eight telephone lines of different types, including:

- Internal POTS/PSTN telephone hybrids
- BT calls
- External TELCO hybrids
- VoIP lines

Instead of assigning every caller to an individual physical fader, the system dynamically routes the selected callers to two logical console sources:

- **TALK SHOW 1**
- **TALK SHOW 2**

These are two mixer sources to which the available telephone lines are consolidated through virtual routing.

TALK SHOW via Web Page (Talk Show Interface):

The **TALK SHOW** mode is managed only through the dedicated Web-based Talk Show interface, which is the main control environment for the OXYGEN 3000 Plus call-management system.

This web page is designed to handle the complete workflow of telephone and hybrid communication independently from the physical console surface, while remaining fully integrated with the mixer's audio architecture. Through the Talk Show web interface it is possible to:

- Monitor all active telephone lines in real time
- View caller status (waiting, active, on-air, hold, conference)
- Assign callers to **TALK SHOW 1** or **TALK SHOW 2**
- Manage waiting queues and priority levels
- Move callers between different operational states
- Activate, hold or release callers from the production flow
- Create and manage conference groups
- Handle hybrid and multi-line configurations
- Coordinate multiple operators with different permissions (Technician, Speaker, Receptionist)

- Prepare callers off-air before sending them to air



The web page acts as the **central control hub for all call logic**, while the console handles the **audio mixing and routing layer**. In this architecture:

- The **Talk Show web interface manages the logical call workflow**
- The console assigns and processes the **audio sources (TALK SHOW 1 and TALK SHOW 2)**
- Callers can be dynamically replaced without changing console routing
- Queue management and caller selection are fully software-driven
- Audio remains continuously synchronized with the mixer engine

The Talk Show web interface also provides a dedicated operational view where the producer or call operator can:

- Drag and assign callers directly to TALK SHOW 1 or TALK SHOW 2
- Update caller status in real time
- Control queue progression and prioritization
- Prepare callers for On Air or off-air use
- Monitor active conversations and conference groups
- Coordinate multiple simultaneous call workflows

Important: the Talk Show web interface is the control layer for caller management. Any change performed on the web page is immediately reflected on the console sources **TALK SHOW 1** and **TALK SHOW 2**, ensuring real-time synchronization between software control and audio mixing.

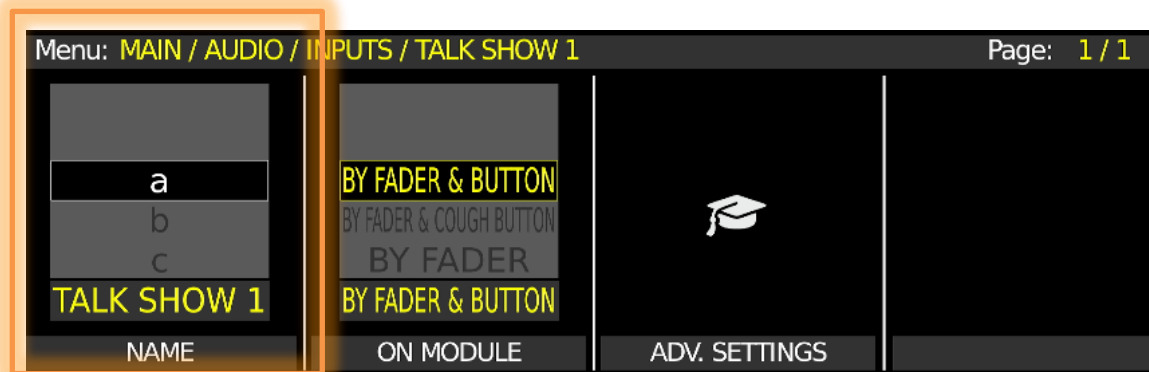
The TALK SHOW input page includes:

- **NAME**
- **ON MODULE**
- **ADV. SETTINGS**

Important: these parameters configure the logical TALK SHOW source inside the console. Individual telephone lines, caller names, waiting queues, conferences and caller assignments are managed separately through the Talk Show Web interface.

INPUTS SETTING (TALK SHOW)

NAME:



NAME assigns a customised identifying label to TALK SHOW 1 or TALK SHOW 2.

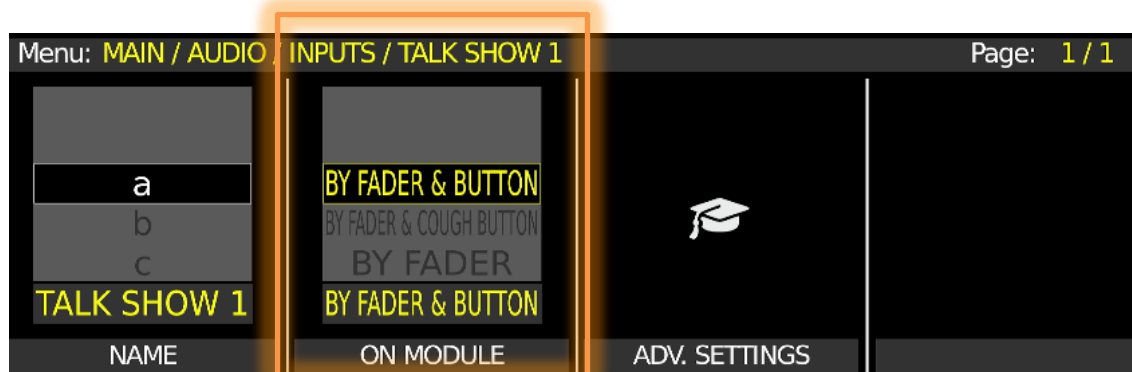
The label can normally contain up to **8 characters**, following the standard input-source naming structure of the OXYGEN 3000 Plus.

Configuration note: the TALK SHOW source name should identify the function of the logical output rather than a specific caller, because the caller assigned to that source may change during operation.

A typical name editing procedure is:

1. Select **NAME**.
2. Rotate the corresponding encoder to choose a character.
3. Press the encoder to confirm the character.
4. Repeat the operation for the remaining positions.
5. Press **ESC** when the label is complete.

ON MODULE:



ON MODULE determines how the channel carrying TALK SHOW 1 or TALK SHOW 2 changes between its logical OFF and ON states.

The selectable options are:

- **FADER & BUTTON:** The TALK SHOW channel is activated through the combined operation of: The physical fader - The ON/START button. The source becomes active when the required button state is enabled and the fader is raised above the configured activation threshold. **Important:** FADER &

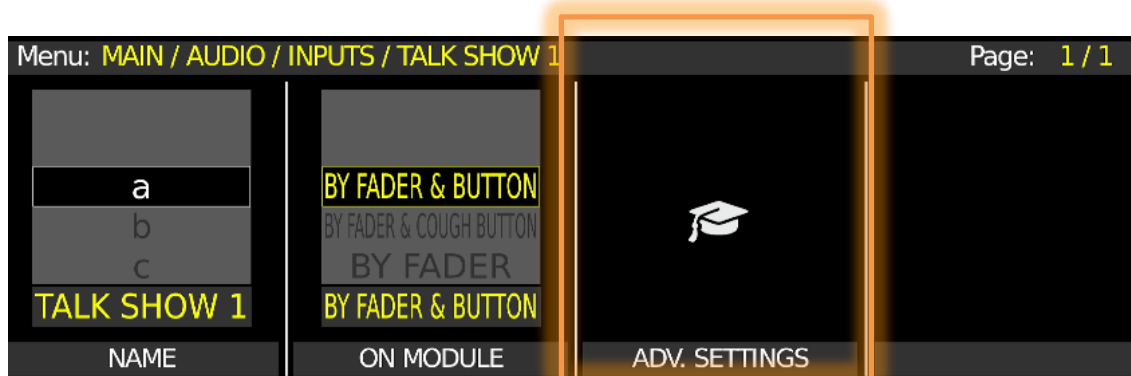
BUTTON controls the console channel, not the caller's telephone connection. Releasing or replacing the caller must still be performed through the Talk Show workflow.

- **BY FADER:** The TALK SHOW channel's logical state follows the physical fader. **Warning:** raising the fader may immediately place the caller assigned to that TALK SHOW source On Air if a programme bus is already selected. Always verify the current caller assignment before opening the fader.
- **BY BUTTON:** The logical TALK SHOW channel state is controlled by the ON/START button. **Important:** closing the fader does not necessarily switch the channel OFF. Always verify the ON/START indication before another caller is assigned to the same TALK SHOW source.
- **ALWAYS ON:** The TALK SHOW source remains logically enabled. **Warning:** ALWAYS ON can make a newly assigned caller immediately available to active pre-fader, direct or auxiliary paths. Verify the current routing before moving a caller onto the source.
- **ALWAYS OFF:** The TALK SHOW source remains logically disabled. The source configuration is retained, but the channel cannot normally be activated through the standard surface controls. **Important:** ALWAYS OFF does not necessarily disconnect callers. A caller may remain connected or waiting inside the Talk Show System even though the console source cannot be activated.

ON MODULE controls the logical TALK SHOW source inside the mixer. It does not answer, release or move individual callers in the Talk Show System.

Important: a caller may remain connected, waiting or assigned to a TALK SHOW source while the corresponding console channel is OFF. The call-management state and the mixer-channel state are separate.

ADV. SETTINGS:



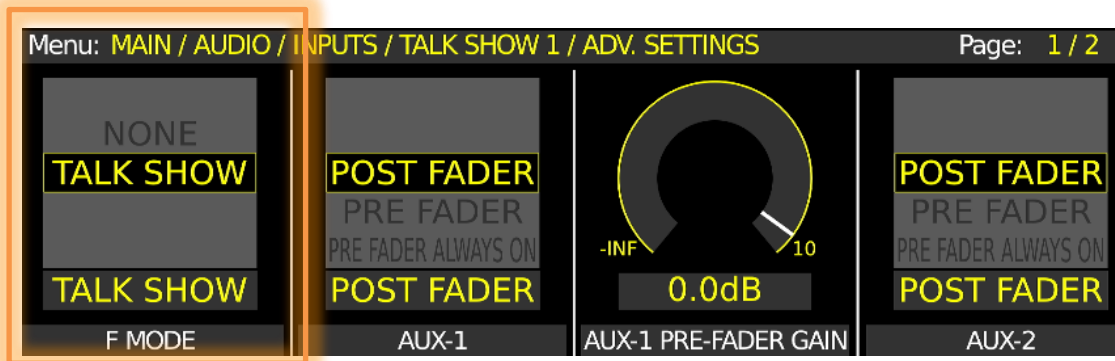
ADV. SETTINGS opens the additional surface-control, auxiliary-routing and illumination parameters assigned to the selected TALK SHOW source.

The section normally includes:

- **F MODE**
- **AUX-1**
- **AUX-1 PRE-FADER GAIN**
- **AUX-2**
- **AUX-2 PRE-FADER GAIN**
- **BUTTON LIGHT**
- **FADER BAR LIGHT**

Important: ADV. SETTINGS configures the resulting TALK SHOW audio source. It does not configure individual telephone lines, waiting queues, conference membership or operator permissions.

F MODE:



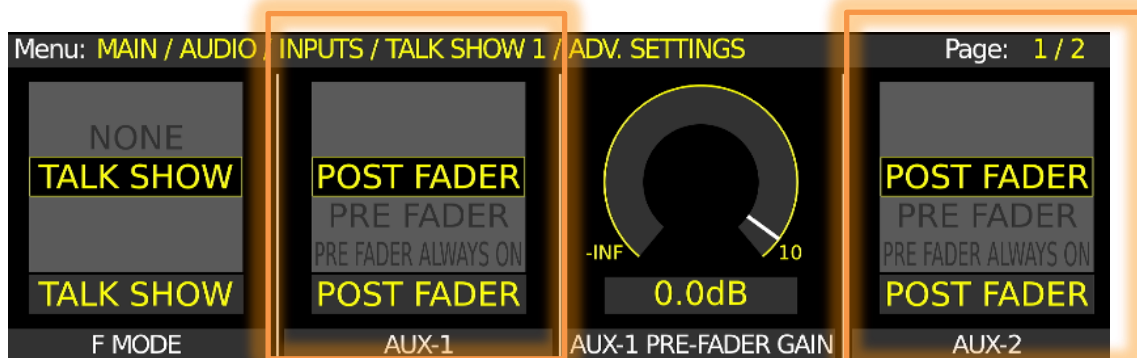
F MODE assigns the Talk Show System function to the programmable F control associated with the channel carrying TALK SHOW 1 or TALK SHOW 2.

The selectable options are normally:

- **NONE:** When F MODE is set to **NONE**, no Talk Show command is assigned to the channel's F control. **Important:** NONE disables only the F-control command. It does not mute TALK SHOW 1 or TALK SHOW 2 and does not disconnect any caller.
- **TALK SHOW MODULE:** It assigns the software-specific control function to the channel's F control. Pressing the control sends a command to the Talk Show System associated with TALK SHOW 1 or TALK SHOW 2.

F MODE determines whether the console surface can send a software-defined command to the Talk Show System.

AUX-1 / AUX-2:



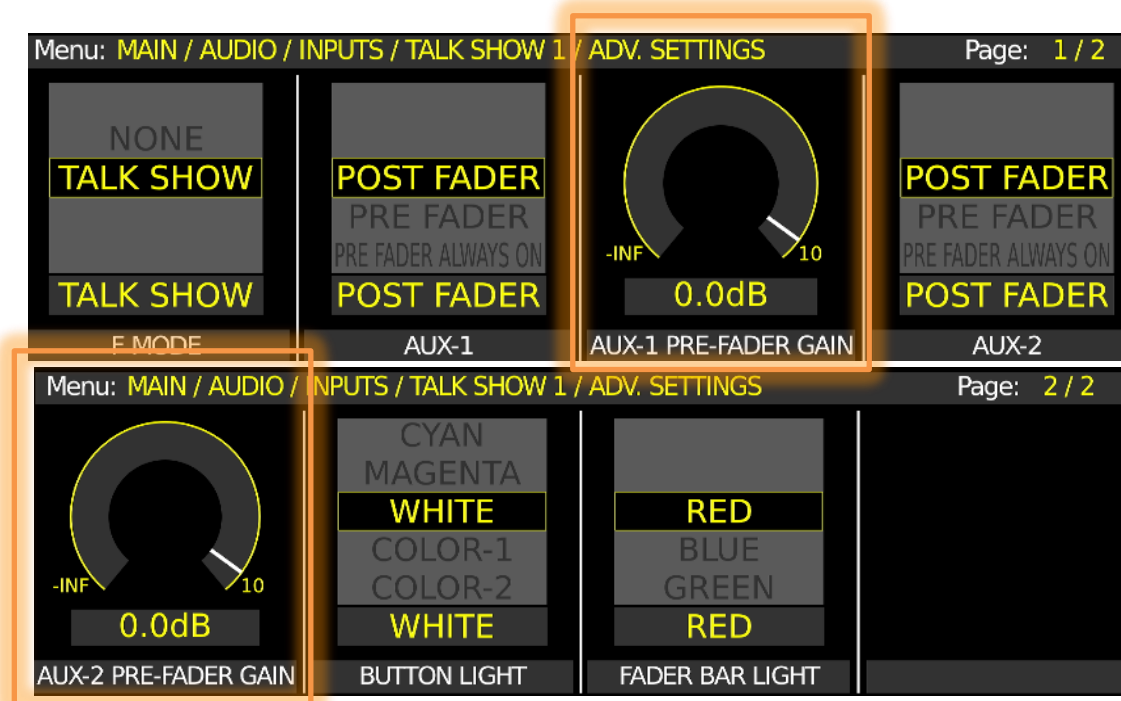
AUX-1 and **AUX-2** independently determine how the selected TALK SHOW source is sent to the corresponding auxiliary bus. The selectable options are:

- **POST FADER:** When AUX-1 or AUX-2 is set to **POST FADER**, the TALK SHOW contribution follows the physical channel fader.

- **PRE-FADER:** When AUX-1 or AUX-2 is set to **PRE-FADER**, the TALK SHOW signal is taken before the physical channel fader. **Important:** PRE-FADER is independent from the physical fader but normally remains subject to the logical channel ON/OFF state.
- **PRE-FADER ALWAYS ON:** When AUX-1 or AUX-2 is set to **PRE-FADER ALWAYS ON**, the TALK SHOW contribution is independent from: The physical channel fader - The ON/START state. **Warning:** PRE-FADER ALWAYS ON can make a private caller audible on an AUX output without an obvious indication from the main fader or ON/START button. A caller newly assigned to the TALK SHOW source may also become audible immediately.

The OXYGEN 3000 Plus supports these three standard AUX operating modes. TALK SHOW 1 and TALK SHOW 2 can use different AUX configurations.

AUX-1 PRE-FADER GAIN / AUX-2 PRE-FADER GAIN:



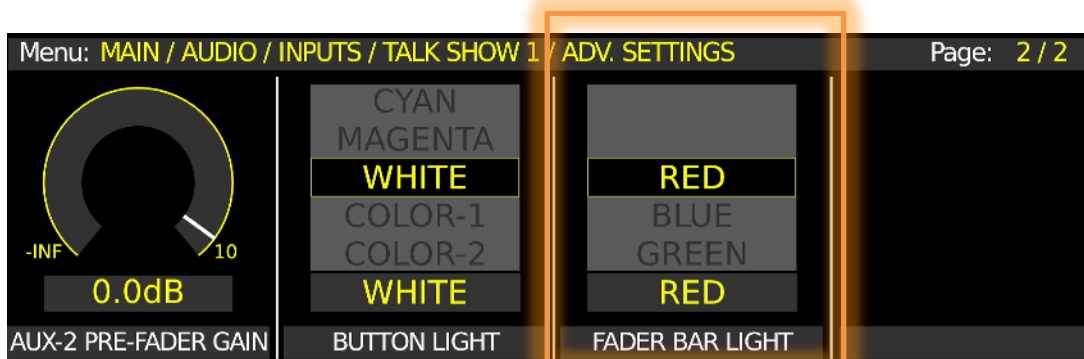
AUX-1 PRE-FADER GAIN and **AUX-2 PRE-FADER GAIN** independently set the TALK SHOW level in the corresponding pre-fader auxiliary path. **Range: -INF to +10.0 dB. Adjustment step: 1.0 dB.**

- Negative values attenuate TALK SHOW in the selected AUX bus.
- **0.0 dB** applies unity gain. The TALK SHOW source is sent to the selected pre-fader AUX path without additional attenuation or amplification.
- Positive values increase the TALK SHOW level in the selected AUX bus.

Warning: excessive positive pre-fader gain may overload the AUX bus, especially when several callers are combined in a conference.

BUTTON LIGHT:

BUTTON LIGHT selects the colour preset used by the illuminated buttons of the channel carrying TALK SHOW 1 or TALK SHOW 2. Each option recalls one of the global RGB colour presets configured in the console.

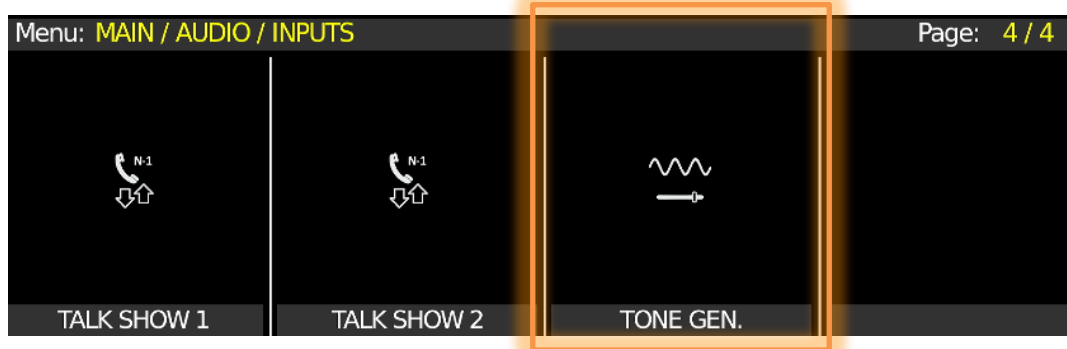
FADER BAR LIGHT:

FADER BAR LIGHT selects the colour preset used by the LED bar associated with the TALK SHOW channel fader. **BUTTON LIGHT** and **FADER BAR LIGHT** are independent.

TONE GEN.

The **TONE GEN** input is an internally generated audio source; therefore, it does not correspond to a physical connector on the console. It produces a continuous test tone that can be assigned to a mixer channel and routed to the **PGM, SUB, AUX 1 or AUX 2 buses** like any other input source.

The tone generator is mainly used for **testing audio connections, checking Left/Right channel routing, calibrating signal levels, verifying meters and outputs, and identifying frequency-response problems**. On the Oxygen 3000 Plus, the tone frequency can be selected within a range from **30 Hz to 20 kHz**.

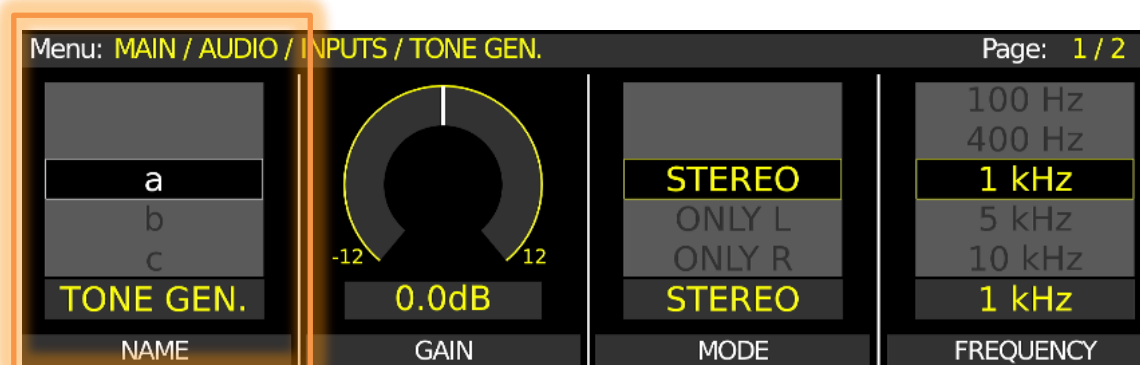


Attention: Before activating the tone generator, reduce the channel and monitor levels. A continuous high-level tone, particularly at very low or high frequencies, may damage loudspeakers or connected equipment.

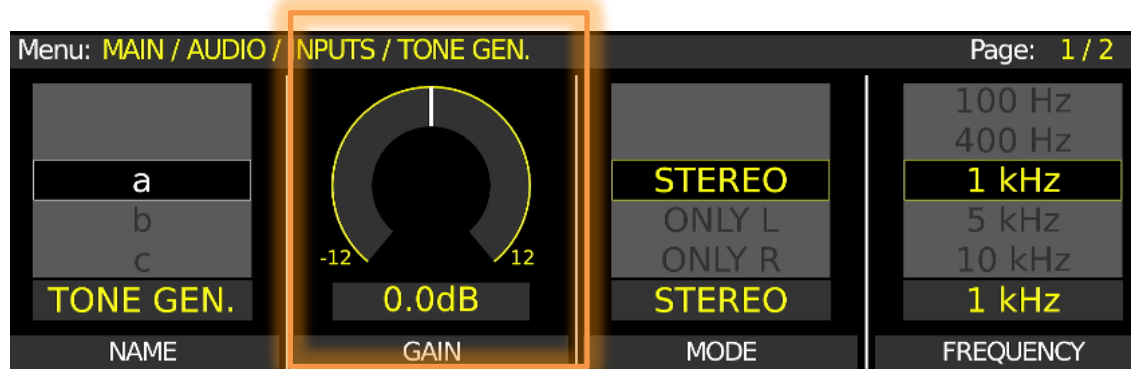
INPUTS SETTING (TONE GEN)

On this page you can modify different parameters: **NAME, GAIN, MODE, FREQUENCY, ON MODULE and ADV SETTINGS**.

NAME:

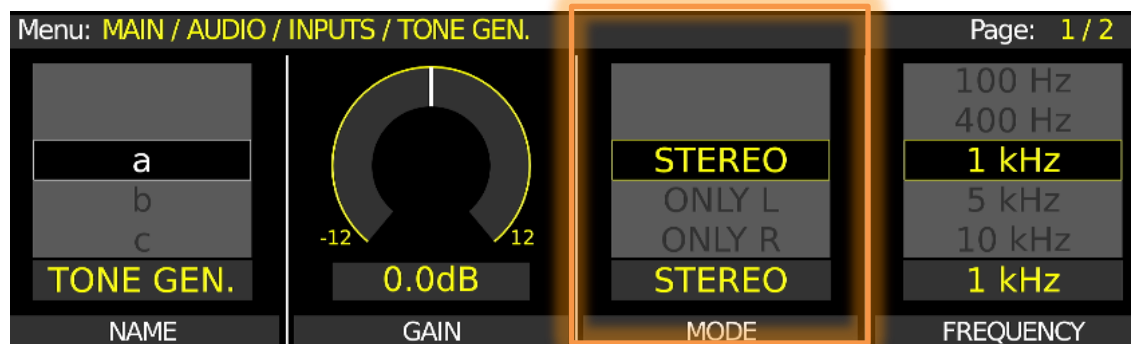


The **Name** parameter assigns a customised label to the Tone Generator source. The selected name is shown on the channel display and helps the operator identify the source when it is assigned to a physical or virtual channel. The name can contain up to **8 characters**. A label such as **TONE, TEST, 1KHZ** or **LINE TEST** may be used according to the purpose of the source.

GAIN:

The **Gain** parameter adjusts the output level of the internally generated tone before it is sent to the channel and the selected audio buses. **Range: -20.0 dB to 20.0 dB. Adjustment step: 0.1**

A lower value is recommended when first activating the source. The level can then be increased gradually while monitoring the console meters.

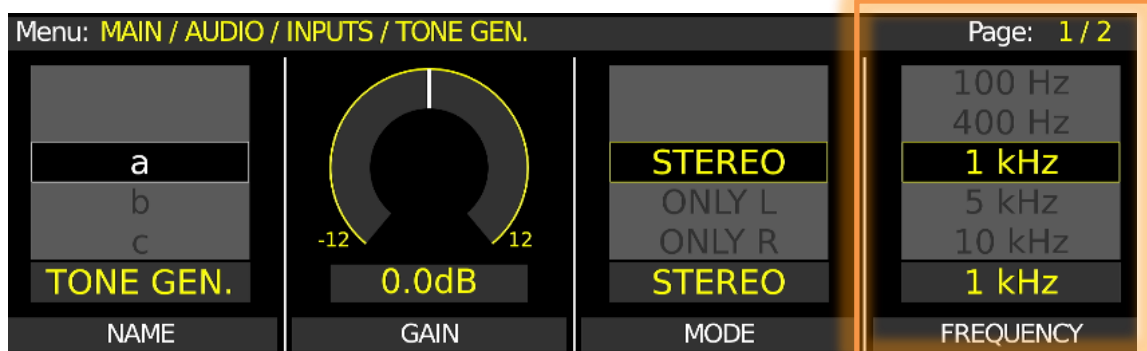
MODE:

The **Mode** parameter determines how the generated tone is distributed between the Left and Right channels. The available options are:

- **STEREO:** sends the test tone simultaneously to both the **Left and Right channels**. This mode is suitable for testing the complete stereo signal path.
- **ONLY L:** sends the test tone exclusively to the **Left channel**, while the Right channel remains silent. It is useful for checking left-channel routing, wiring and output operation.
- **ONLY R:** sends the test tone exclusively to the **Right channel**, while the Left channel remains silent. It is useful for checking right-channel routing, wiring and output operation.

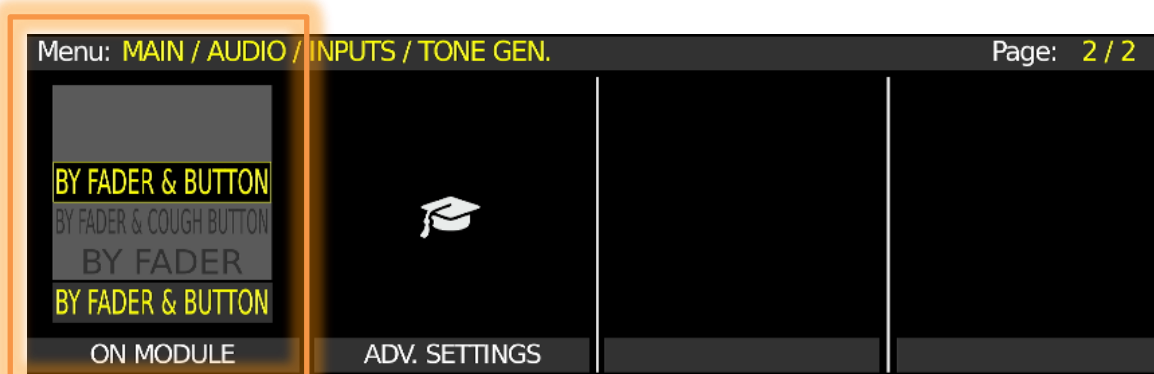
These options allow the operator to test the **entire stereo path or each channel independently**, making it easier to identify incorrect routing, reversed connections or faults affecting only one side of the audio system.

FREQUENCY:



The **Frequency** parameter sets the frequency of the generated test tone. The selectable range extends from **30 Hz to 20 kHz**. The selected frequency remains constant until the value is changed or the Tone Generator source is switched off.

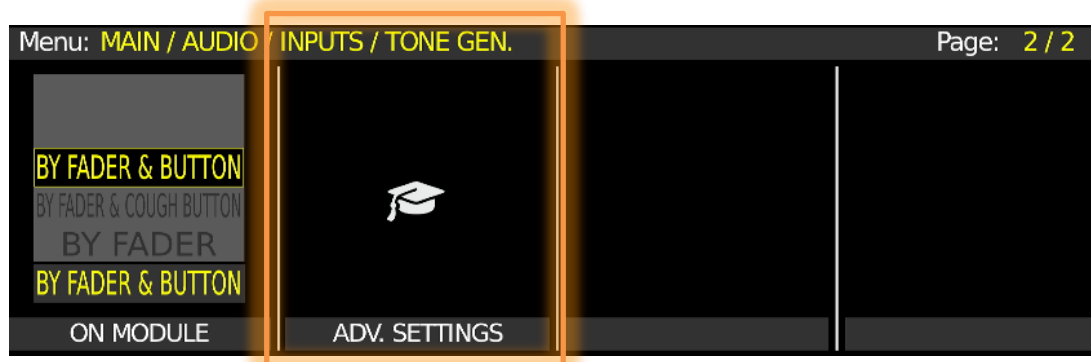
ON MODULE:



The **On Module** parameter determines how the Tone Generator channel is opened or closed from the console surface. The available options can include:

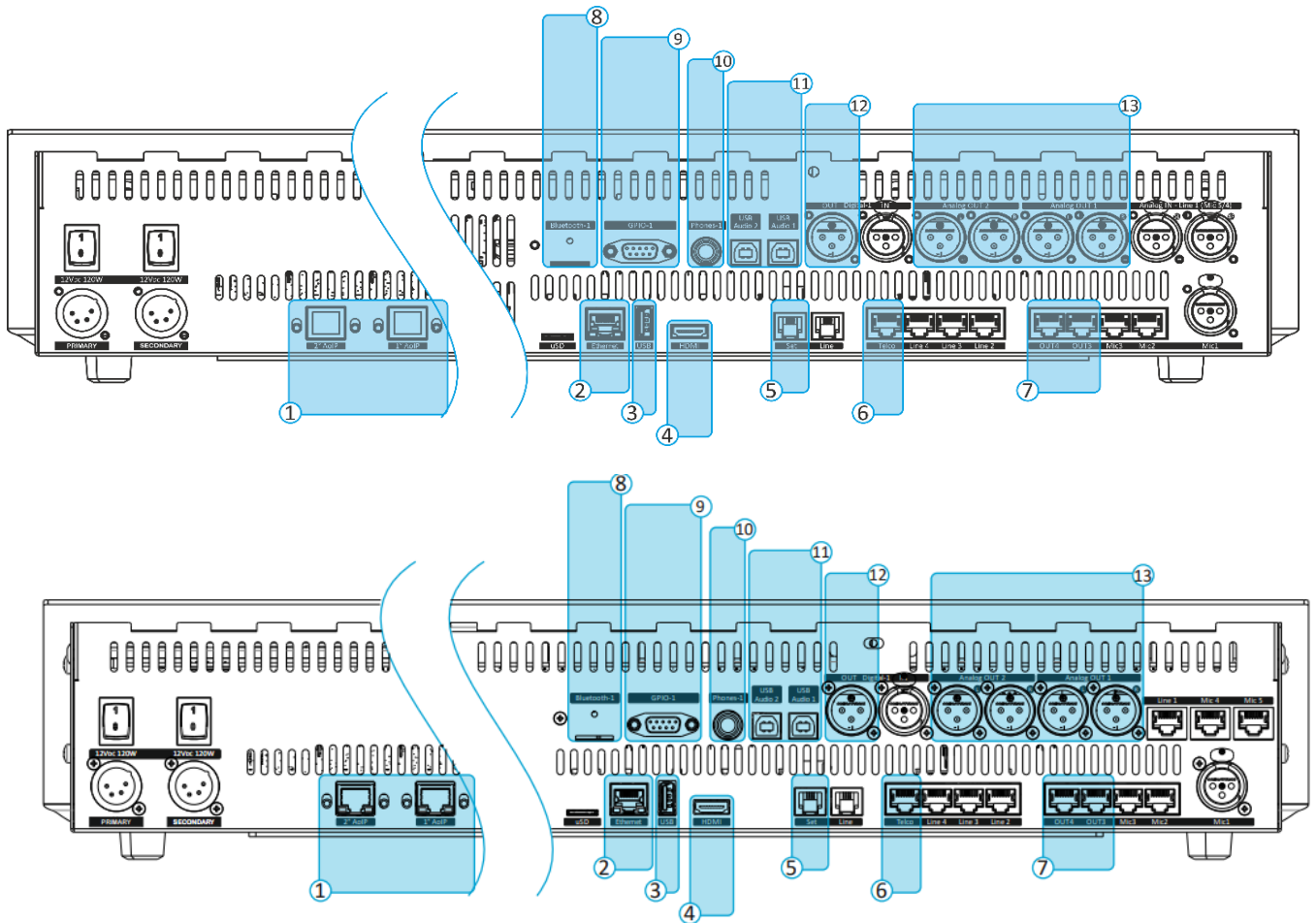
- **FADER & BUTTON:** the channel must be enabled with the **ON/START button**, and its level is controlled by the fader.
- **BY FADER:** raising the fader opens the channel; lowering it below the activation threshold closes the channel.
- **BY BUTTON:** the channel is opened or closed exclusively through the **ON/START button**. The fader controls only the output level.
- **BY FADER & COUGH BUTTON:** the fader controls normal activation, while a configured Cough button can temporarily mute the source. This option is relevant only when a compatible external control is available.
- **ALWAYS ON:** the source remains enabled independently of the fader and ON/START button. The tone may therefore remain present in an assigned bus.
- **ALWAYS OFF:** the source is permanently disabled and cannot be opened from the channel controls.

For normal testing, **BY BUTTON** or **FADER & BUTTON** is recommended because these modes prevent the test tone from being activated accidentally.

ADV. SETTINGS:

The **Advanced Settings** menu contains only the personalization of the **BUTTON LIGHT** and the **FADER BAR LIGHT**.

3. OUTPUTS

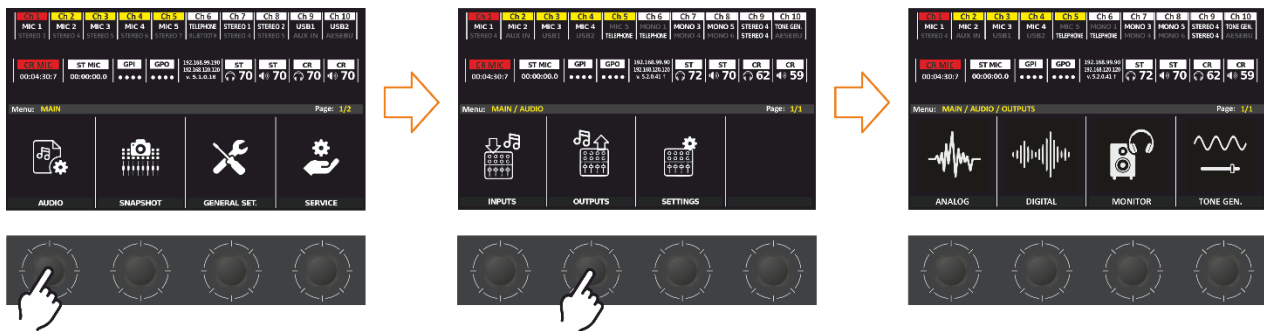


OXYGEN Digital console is coming with variant audio output connectors like **XLR, JACK, USB, and RJ45** to give you maximum flexibility with your instruments and your project cabling. With all those functions you can save time and money. As well as the creative USB connection and Dante technology, we don't need a sound card or a professional instrument to connect any media player or recorder like a **PC, Workstation, or laptop**. **Unlike the input sources, all outputs can be selected and used simultaneously, without requiring any switch configuration in the settings.**

OXYGEN outputs included:

- **2 Stereo XLR Male** - Balanced Audio Connection (**47Ω**).
- **2 Stereo RJ45 Female** audio and **GPIO** connections.
- **1 Telco I/O RJ45 Female** audio and **GPIO** connections.
- **1 Analog Telephone Line - RJ11 - PSTN** Interface
- **1 AES/EBU Stereo XLR Male** - Balanced Digital Connection (**110Ω**).
- **32 Stereo RJ45 Female** DANTE audio.
- **1 USB (USB2) Audio Card USB -Type B** - PC Connections.
- **1 Stereo Output - Jack 6.3mm** - Unbalanced Audio Connection (**nominal 32Ω**).

You can set output parameters from the **MENU: MAIN / AUDIO / OUTPUTS** by **PUSH/ROTATE** the knobs as in the following steps.



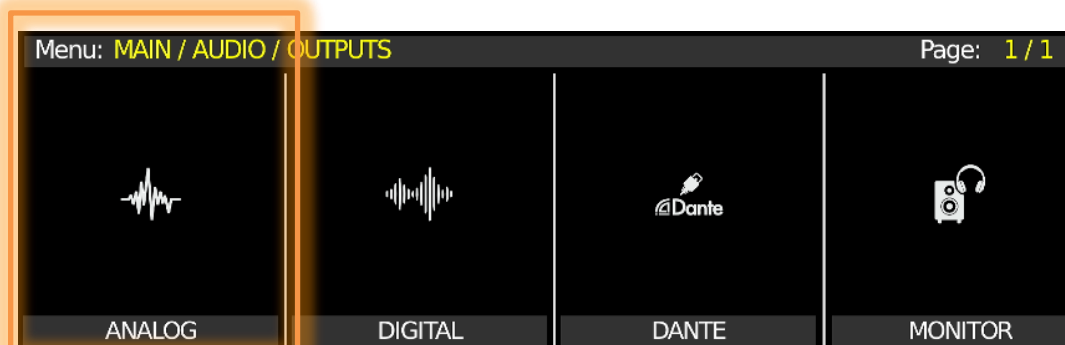
From this level of the menu, you choose which output you like to customize.



ATTENTION: In all **OUTPUT** section, the availability of **ADVANCED SETTINGS** depends on the selected **SOURCE**. If the source is set to **PGM, SUB, AUX 1, AUX 2**, or any **TELCO** option, **ADVANCED SETTINGS** can be selected. If the source is set to **HDP-CRM, SPK-CRM, HDP-STD, or SPK-STD**, **ADVANCED SETTINGS** are not available.

3.1 ANALOG OUTPUTS

The **ANALOG** category includes the physical connections used to send audio from the Oxygen 3000 Plus to external analogue equipment. The console converts its internally processed digital audio into an analogue electrical signal through its D/A converters. The analogue line outputs can be assigned to different mixer signals, including **PGM, SUB, AUX 1, AUX 2, speaker monitoring, headphone monitoring and Telco N-1**. This makes them suitable for feeding broadcast processors, active monitors, amplifiers, recorders, telephone hybrids and headphone distribution systems.



ANALOG 1-2 — XLR:

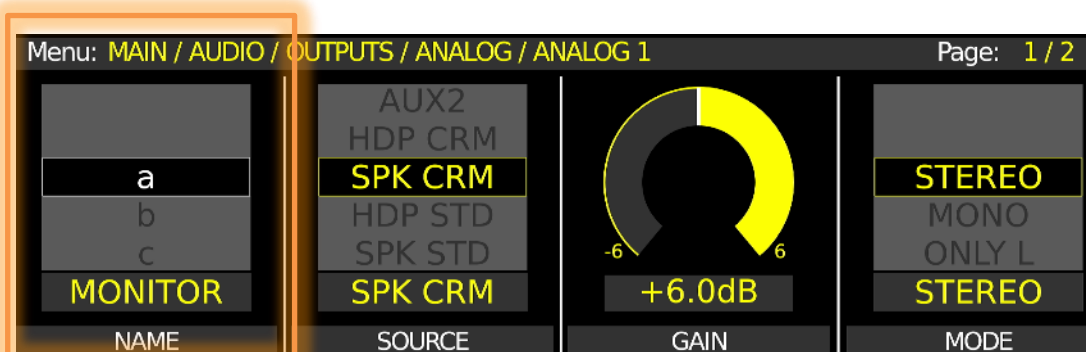
ANALOG 1 and **ANALOG 2** are independently configurable **balanced stereo line outputs** provided through male XLR connectors.



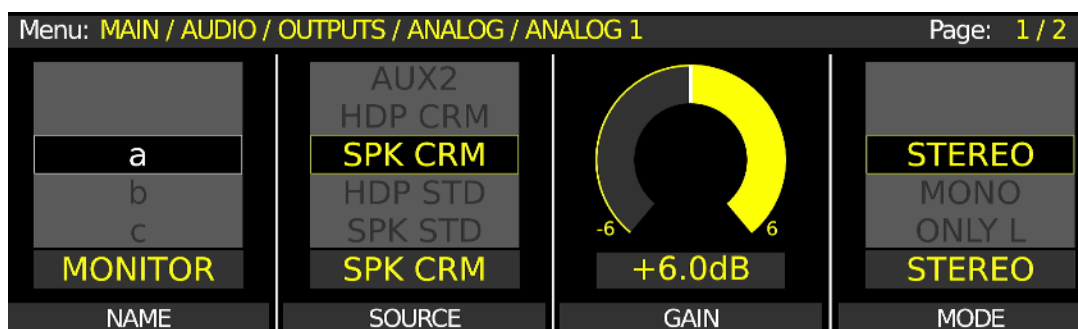
Their parameters determine which internal signal is routed to the physical output, its level and the way its Left and Right channels are reproduced. In this page you can set different options: **NAME SOURCE, GAIN, MODE, ADV SETTINGS**.

INPUTS SETTING (ANALOG 1-2 (XLR))

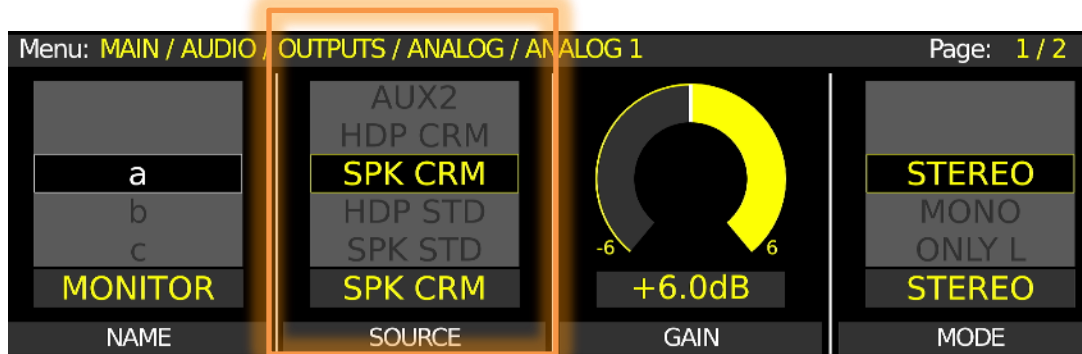
NAME:



Assigns a customised name to the selected output, making it easier to identify within the console interface. The label can contain up to **8 characters**. Changing the name has **no effect on the audio signal**.



SOURCE:



Selects the internal audio signal routed to the selected ANALOG 1 or ANALOG 2 XLR output.

- **PGM** – Routes the main **Programme bus**. This is normally used for the primary broadcast, transmission or programme-recording chain.
- **SUB** – Routes the **Sub bus**, providing a secondary or alternative programme mix.
- **AUX 1** – Routes the first auxiliary bus. It can be used for independent monitoring, recording, external processing or communication feeds.
- **AUX 2** – Routes the second auxiliary bus, providing an additional independent mix.
- **HDP CRM** – Routes the **Control Room headphone-monitoring signal**. The output reproduces the source and processing selected for the Control Room headphone path.
- **SPK CRM** – Routes the **Control Room loudspeaker-monitoring signal**.
- **HDP STD** – Routes the **Studio headphone-monitoring signal**.
- **SPK STD** – Routes the **Studio loudspeaker-monitoring signal**.
- **HDP CRM 90%** – Routes the Control Room headphone-monitoring signal using the console's dedicated **90% level reference**.
- **SPK CRM 90%** – Routes the Control Room loudspeaker-monitoring signal using the dedicated **90% level reference**.
- **HDP STD 90%** – Routes the Studio headphone-monitoring signal using the dedicated **90% level reference**.
- **SPK STD 90%** – Routes the Studio loudspeaker-monitoring signal using the dedicated **90% level reference**.
- **DIRECT OUT:** Routes a selected input source directly to the Dante output without first using one of the main mixing buses, the source is selectable in advanced settings sections.

Important: the **90% designation refers to an internal percentage-based level reference**. It must not be interpreted as a –10 dB attenuation or as 90 dB. The final physical output level can still be adjusted using the output parameter. You can select as source from Telco 3 to Telco 10. Each option routes the **N–1 or mix-minus signal** associated with the selected Telco channel. The return feed contains the required programme or communication signals while excluding the corresponding caller's own incoming audio.

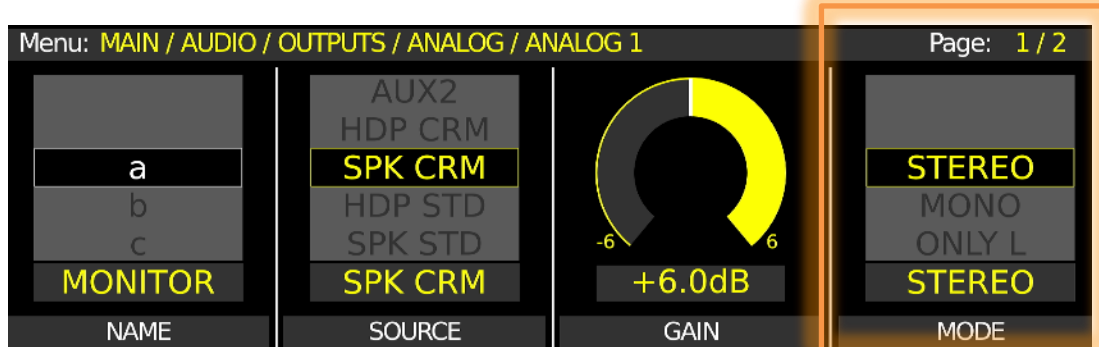
GAIN:



Adjusts the final level delivered by the selected analogue output. **Range: -6.0 dB to +6.0 dB. Adjustment step: 0.1 dB** At **0.0 dB**, the signal is delivered at unity gain. Negative values attenuate the output, while positive values increase it. The adjustment is applied in real time. The level should be set carefully to avoid clipping the input of the connected equipment.

N.B.: When the parameter control is activated, in real-time the parameter value is updated, perceiving the increase and decrease of the **GAIN**, the graphics will change from yellow to white until the parameter is confirmed.

MODE:



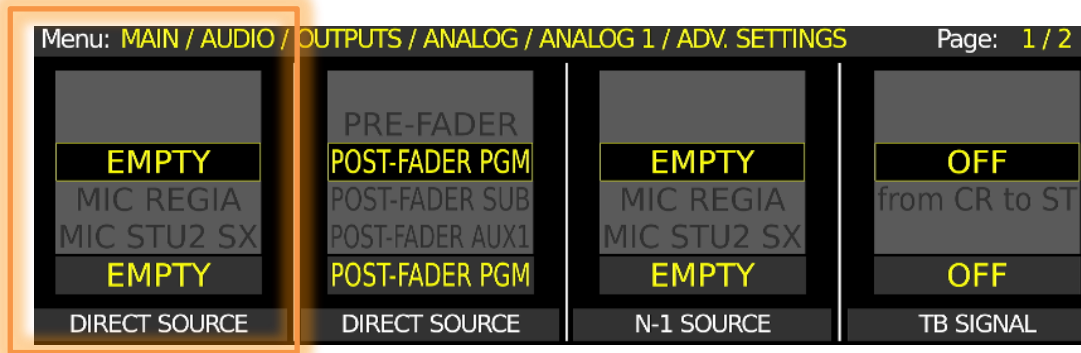
Determines how the Left and Right channels are reproduced at the physical output.

- **STEREO** – Preserves the original Left and Right channels as an independent stereo pair.
- **MONO** – Combines the Left and Right channels and sends the resulting mono mix to both output channels.
- **MONO LEFT** – Uses only the Left channel and reproduces it as a mono signal on both output channels.
- **MONO RIGHT** – Uses only the Right channel and reproduces it as a mono signal on both output channels.

ADV. SETTINGS:

Opens the secondary configuration page for additional output-specific parameters. In ADV. SETTINGS you can set different parameters: **DIRECT SOURCE**— **Source selection**, **DIRECT SOURCE** — **Signal pick-off point**, **N-1 SOURCE**, **TB SIGNAL**, **TB ACTIVATION**.

DIRECT SOURCE — Source selection:

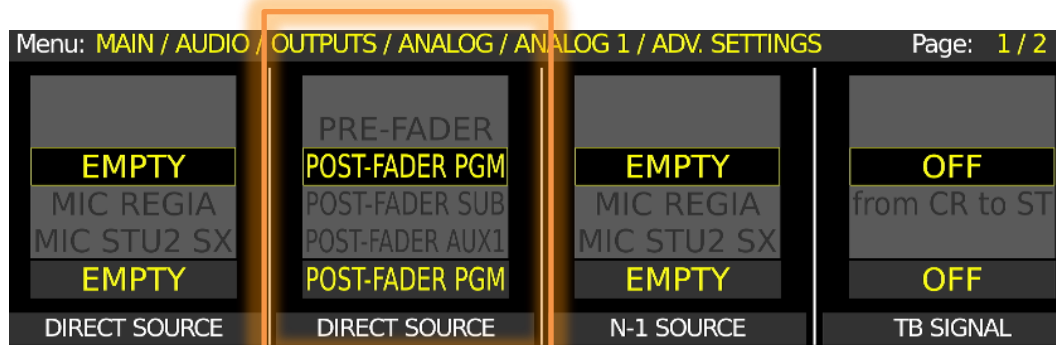


Selects the input that will be routed directly to the analogue output. Available selections include:

- **EMPTY** – Disables the Direct Source path.
- **Any available Oxygen 3000 Plus input source** – Selects the required input and routes it directly to the output.

The list therefore includes all input sources currently enabled in the console, such as microphones, Mono and Stereo inputs, AES/EBU, USB, Dante, Bluetooth, Streaming, telephone sources, Telco sources, VoIP, Talk Show, Tone Generator and other available internal sources.

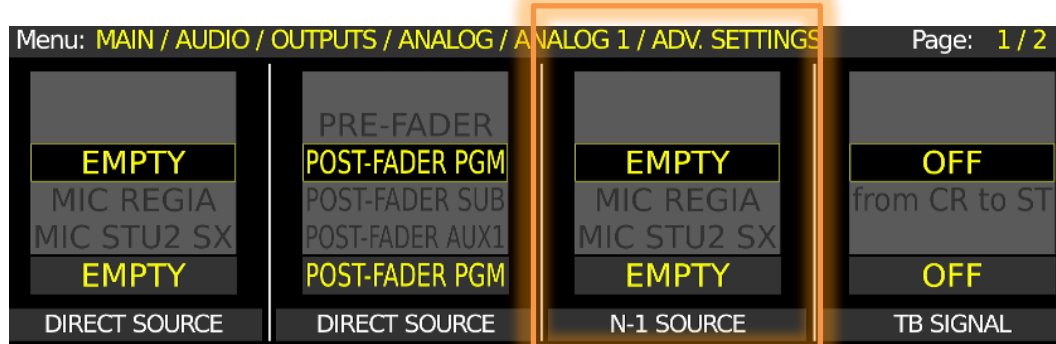
DIRECT SOURCE — Signal pick-off point



Determines where the selected Direct Source is taken from the channel's signal path.

- **PRE FADER** – Takes the signal before the channel fader. The output level is therefore **independent of fader movements**.
- **POST FADER PGM** – Takes the selected source from its post-fader PGM path. The signal follows the channel fader and its PGM routing condition.
- **POST FADER SUB** – Takes the signal from the post-fader SUB path.
- **POST FADER AUX 1** – Takes the signal from the post-fader AUX 1 path.
- **POST FADER AUX 2** – Takes the signal from the post-fader AUX 2 path.

N-1 SOURCE

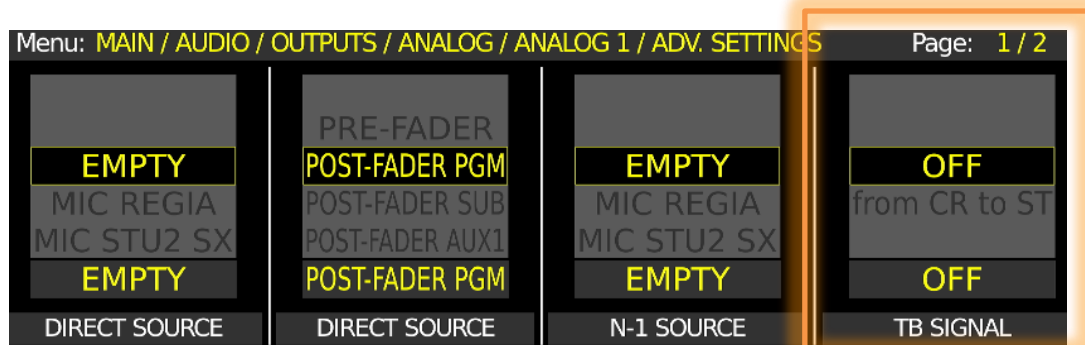


Selects the input source that must be removed from the outgoing signal to create an **N-1 or mix-minus feed**. All available Oxygen 3000 Plus input sources can be selected. The list changes according to the sources enabled in the current console configuration. When an input is selected as the N-1 SOURCE:

- The selected source is **removed from its own return feed**.
- The remaining required programme or communication signals continue to be sent.
- The connected participant does not receive a delayed copy of their own audio.

Although N-1 is most commonly used with **Telco, telephone, VoIP, Bluetooth and remote-codec sources**, it can be applied to any available input when a customised mix-minus feed is required.

TB SIGNAL

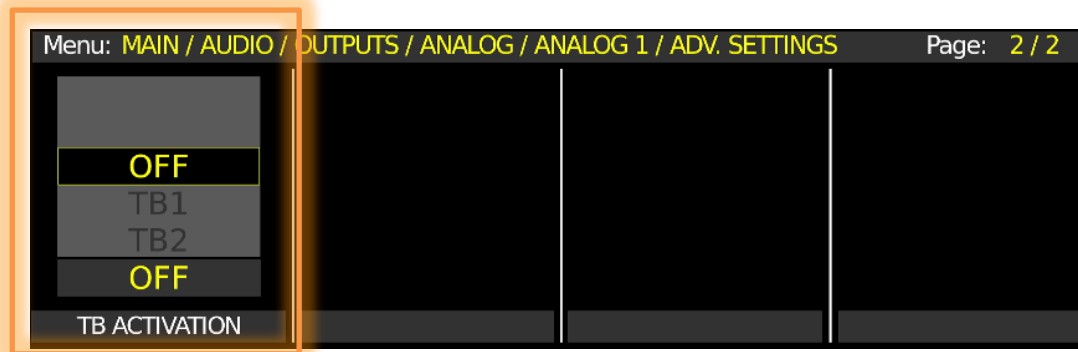


Determines whether the **Control Room-to-Studio Talkback signal** is included in the selected analogue output. The available options are:

- **OFF** – The Control Room-to-Studio Talkback signal is not sent to the output.
- **FROM CR TO ST** – Routes the Talkback communication generated in the **Control Room and directed to the Studio** to the selected analogue output.

This parameter does not select the Talkback microphone. The Talkback audio source is determined by the console's Control Room Talkback configuration.

TB ACTIVATION



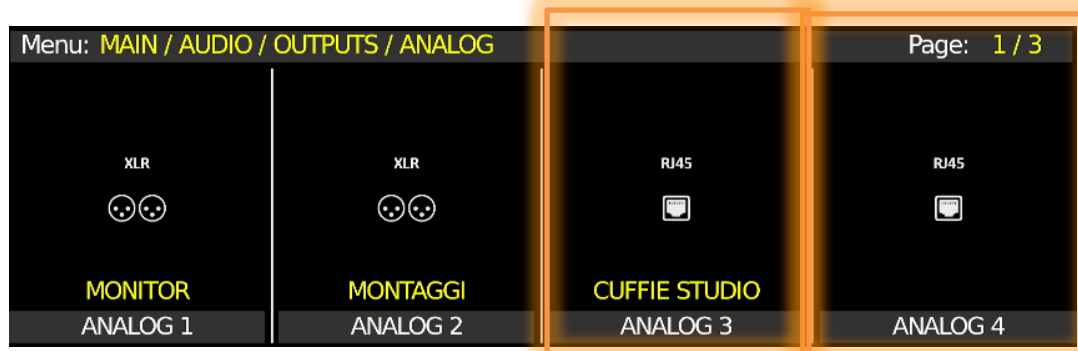
Selects the Talkback command that activates the **Control Room-to-Studio Talkback signal** on the selected ANALOG 1 or ANALOG 2 output. The available options are:

- **OFF** – Disables Talkback activation on the output. No Talkback signal is sent when the TB buttons are pressed.
- **TB1** – Activates the Talkback signal when the **TB1 command** is operated.
- **TB2** – Activates the Talkback signal when the **TB2 command** is operated.
- **TB3** – Activates the Talkback signal when the **TB3 command** is operated.
- **TB4** – Activates the Talkback signal when the **TB4 command** is operated.

TB SIGNAL selects the Talkback audio path, while TB ACTIVATION selects the Talkback command that enables it.

ANALOG 3–4 — RJ45:

ANALOG 3 and **ANALOG 4** are assignable **balanced stereo analogue outputs** provided through RJ45 connectors. They offer the same principal routing functions as ANALOG 1–2 but use a compact connector that simplifies installation in structured studio wiring systems. **Important:** although the connector is physically an RJ45, ANALOG 3–4 are **not Ethernet or Dante network ports**.



The **OUTPUT SETTINGS** for **ANALOG 3–4 (RJ45)** are identical to those previously described for **ANALOG 1–2 (XLR)**. The only exception is that **ADVANCED SETTINGS** are not available in this section.

HEADPHONES 1:

HEADPHONES 1 is an unbalanced stereo monitoring output provided through a **6.3 mm TRS jack connector**. It is designed primarily for connecting the **Control Room headphones** directly to the console. It allows the operator to monitor the selected Control Room source, such as **PGM, SUB, AUX 1, AUX 2** or **an external monitoring source**. Depending on the monitor configuration, it can also reproduce **PFL** and **Talkback signals**. The output is intended for headphones with a **nominal impedance of 32 Ω or higher**. Unlike ANALOG 1–4, HEADPHONES 1 is a dedicated headphone-monitoring connection and should not normally be used as the main feed for professional balanced line equipment.



The **OUTPUT SETTINGS** for **HEADPHONES 1** are identical to those previously described for **ANALOG 1–2 (XLR)**, except for the **MODE** section. In addition, **ADVANCED SETTINGS** are not available for this output.

3.2 DIGITAL OUTPUTS

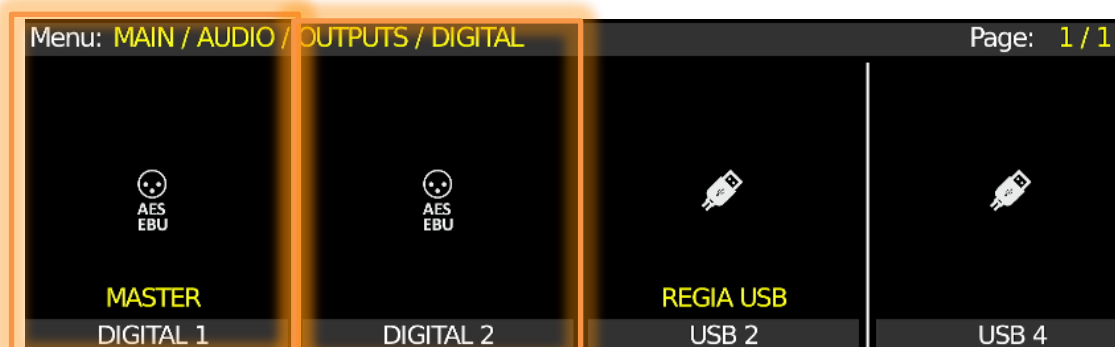
A **digital output** sends audio from the Oxygen 3000 Plus as numerical audio data rather than as a continuously varying analogue voltage. Because the signal remains in the digital domain, it can be transferred to compatible devices without an additional digital-to-analogue conversion.



DIGITAL 1 (AES/EBU):

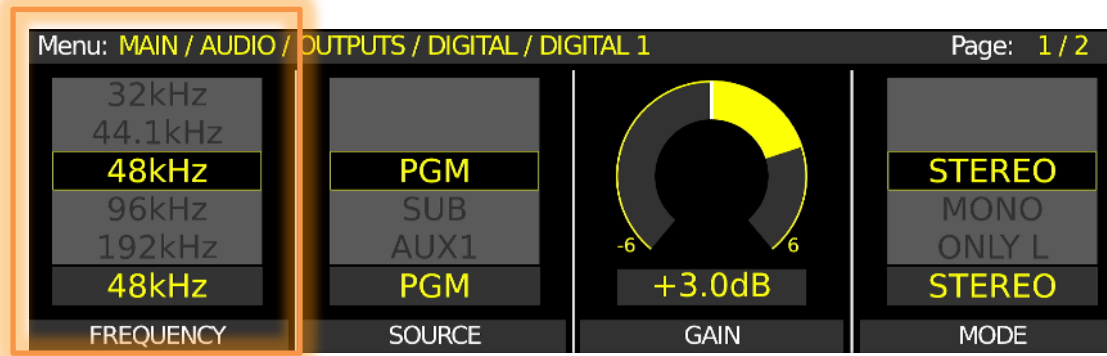
DIGITAL 1 is the professional stereo digital output of the Oxygen 3000 Plus. It uses a **male XLR connector**, a balanced **110 Ω AES3/AES/EBU connection** and a **24-bit digital audio format**.

Unlike an analogue output, it transmits both the audio data and the related digital clock. It can be connected to compatible broadcast processors, digital recorders, codecs, audio interfaces, transmitters and digital routing systems.



INPUTS SETTING (DIGITAL 1 (AES/EBU))

FREQUENCY:



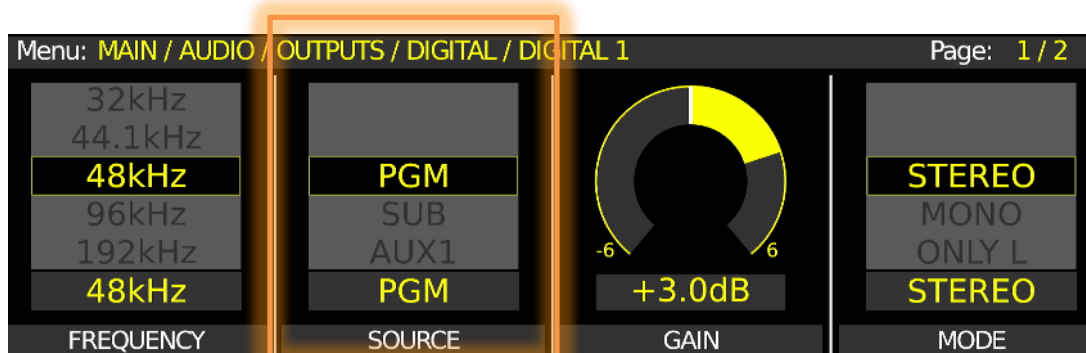
Selects the sampling frequency used by the AES/EBU output. Available options are:

- **32 kHz** – Suitable for compatible legacy or speech-oriented digital equipment.
- **44.1 kHz** – Commonly used by music and audio-production systems.
- **48 kHz** – Standard frequency for professional broadcast, video and studio applications.
- **96 kHz** – Provides a higher sampling frequency for compatible professional equipment.
- **192 kHz** – Provides the highest supported sampling frequency and requires a compatible receiving device.

Important: the transmitter and receiver must operate at the same sampling frequency. An incompatible setting may result in **no audio, clicks, interruptions or an unlocked digital input**.

Changing FREQUENCY does not select a different audio source and does not change the output's nominal **24-bit resolution**.

SOURCE:



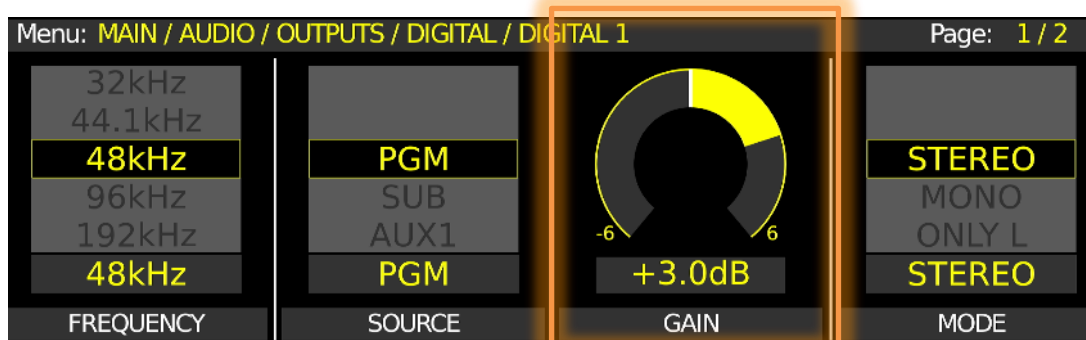
Selects the internal mixer signal routed to DIGITAL 1. Available selections include:

- **PGM** – Main Programme bus.
- **SUB** – Secondary programme bus.
- **AUX 1** – First auxiliary bus.
- **AUX 2** – Second auxiliary bus.
- **HDP CRM** – Control Room headphone-monitoring signal.
- **SPK CRM** – Control Room loudspeaker-monitoring signal.

- **HDP STD** – Studio headphone-monitoring signal.
- **SPK STD** – Studio loudspeaker-monitoring signal.
- **HDP CRM 90%** – 90% level variant of the Control Room headphone signal.
- **SPK CRM 90%** – 90% level variant of the Control Room loudspeaker signal.
- **HDP STD 90%** – 90% level variant of the Studio headphone signal.
- **SPK STD 90%** – 90% level variant of the Studio loudspeaker signal.
- **N-1 TELCO 3 to N-1 TELCO 10** – Mix-minus return associated with the selected Telco channel.
- **DIRECT OUT:** Routes a selected input source directly to the Dante output without first using one of the main mixing buses, the source is selectable in advanced settings sections.

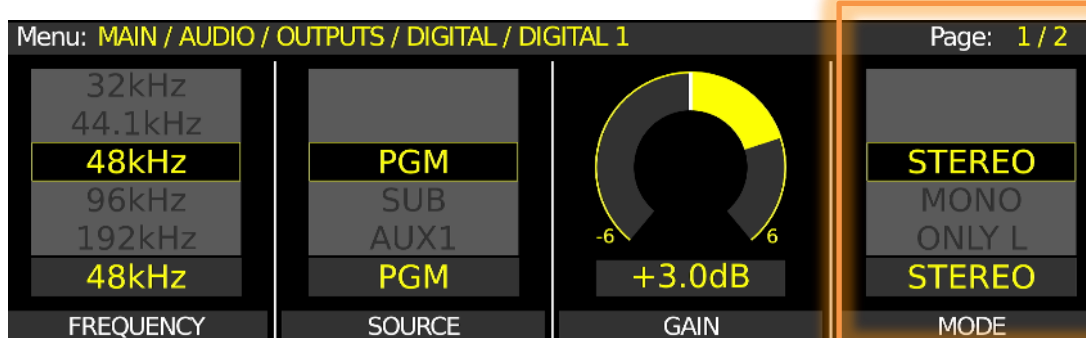
The N-1 options exclude the corresponding Telco input from its return signal, preventing the caller from receiving a delayed copy of their own voice. The **90% designation** identifies the firmware-defined 90% variant of the corresponding monitor signal. It must not be interpreted as **-10 dB** or **90 dB**.

GAIN:



Adjusts the digital level delivered by the AES/EBU output. **Range: -6.0 dB to +6.0 Db. Adjustment step: 0.1 dB**. Negative values attenuate the digital signal, while positive values increase it. Excessive positive gain may produce **digital clipping**, especially if the selected source is already close to 0 dBFS.

MODE:

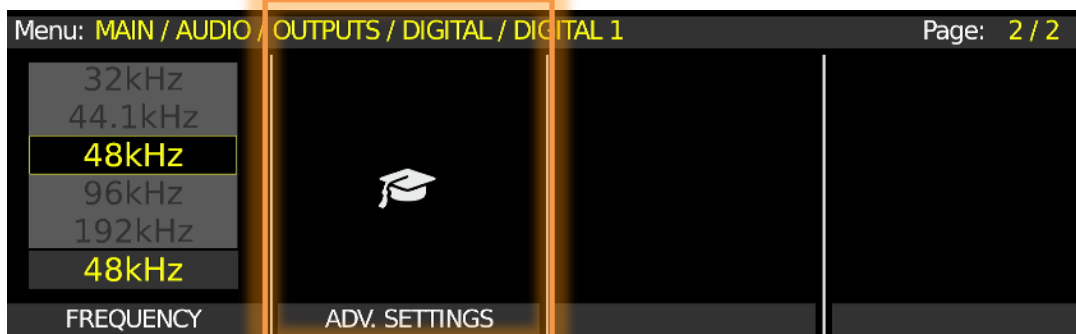


Determines how the Left and Right channels are transmitted through the AES/EBU output.

- **STEREO** – Preserves the original independent Left and Right channels.
- **MONO** – Combines the Left and Right channels and sends the resulting mono signal on both AES/EBU channels.
- **MONO LEFT** – Uses only the original Left channel and sends it as a mono signal on both channels.

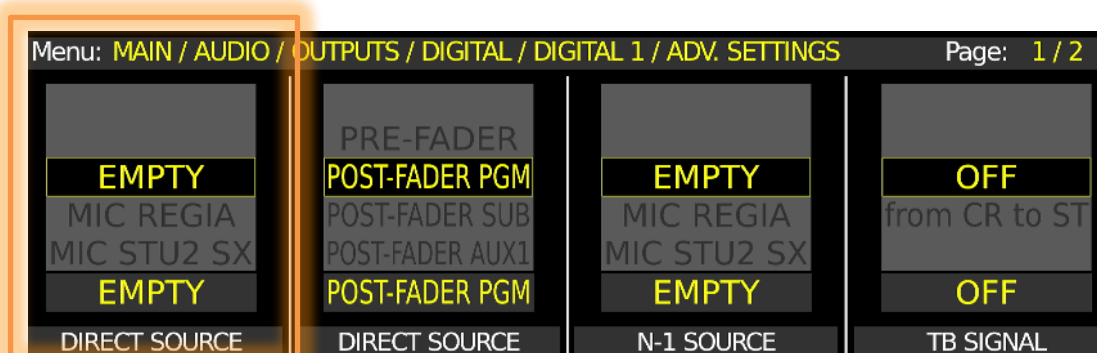
- **MONO RIGHT** – Uses only the original Right channel and sends it as a mono signal on both channels.

ADV. SETTINGS



Provides additional routing, N–1 and Talkback controls for DIGITAL 1. The **ADVANCED SETTINGS** for **DIGITAL 1** are essentially identical to those previously described for **ANALOG 1–2 (XLR)**. For further details beyond the information provided in the following sections, refer to the **ANALOG 1–2 (XLR)** section under **ANALOG OUTPUTS**.

DIRECT SOURCE — Source selection

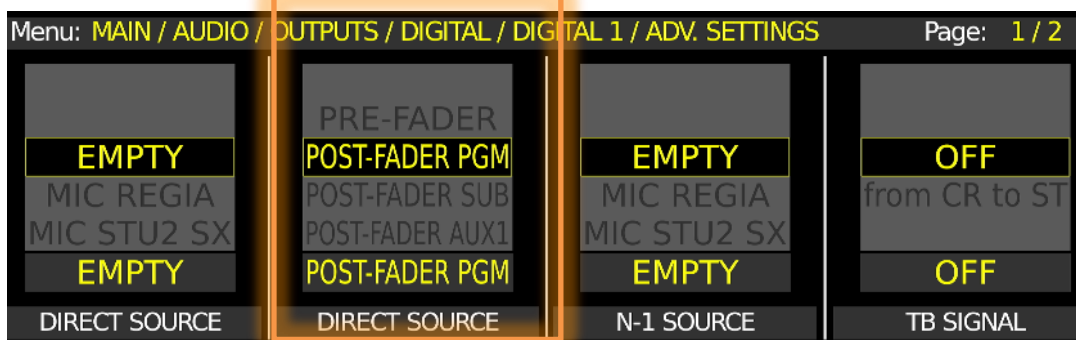


Selects an individual input to be routed directly to the digital output.

- **EMPTY** – Disables direct source routing.
- **Any available Oxygen 3000 Plus input** – Routes the selected input directly to DIGITAL 1.

The list includes all sources enabled by the current hardware and INPUT MODE configuration.

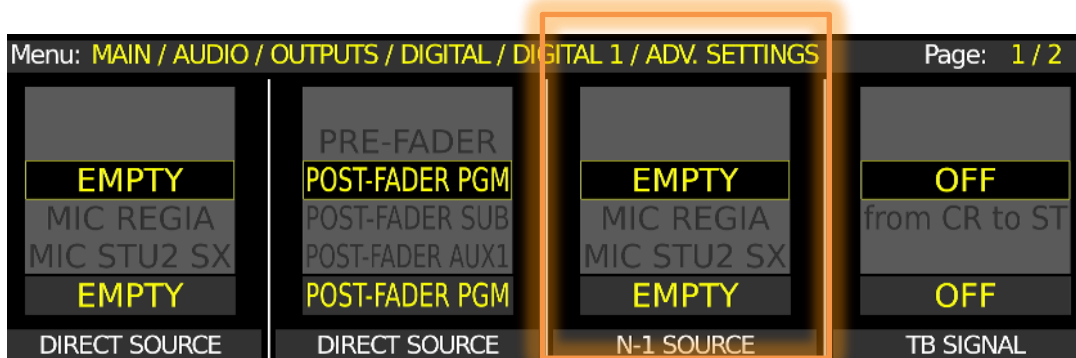
DIRECT SOURCE — Pick-off point



Selects the point from which the direct signal is taken.

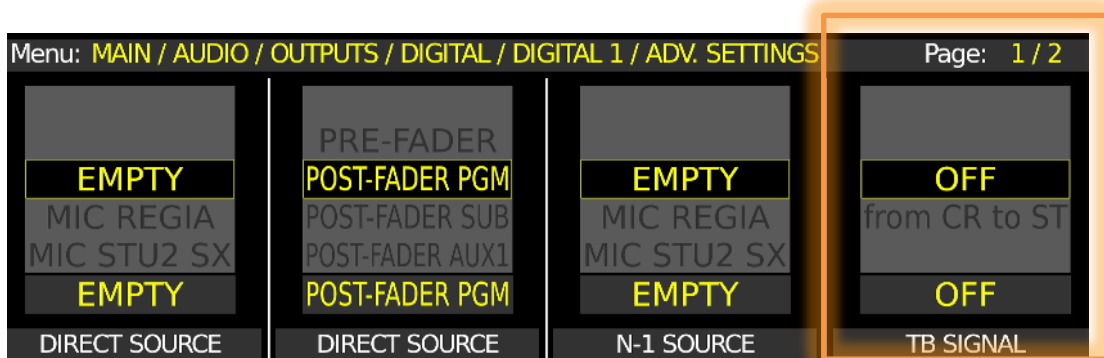
- **PRE FADER** – The direct output is independent of the channel fader.
- **POST FADER PGM** – The output follows the channel fader and its PGM routing.
- **Other POST FADER bus options** – Use the equivalent post-fader path displayed by the console.

N-1 SOURCE



Selects the input that must be removed from the outgoing mix. **Any available Oxygen 3000 Plus input source** can be selected. This function creates a customised **mix-minus feed** for telephone, VoIP, codec or remote-contribution applications.

TB SIGNAL

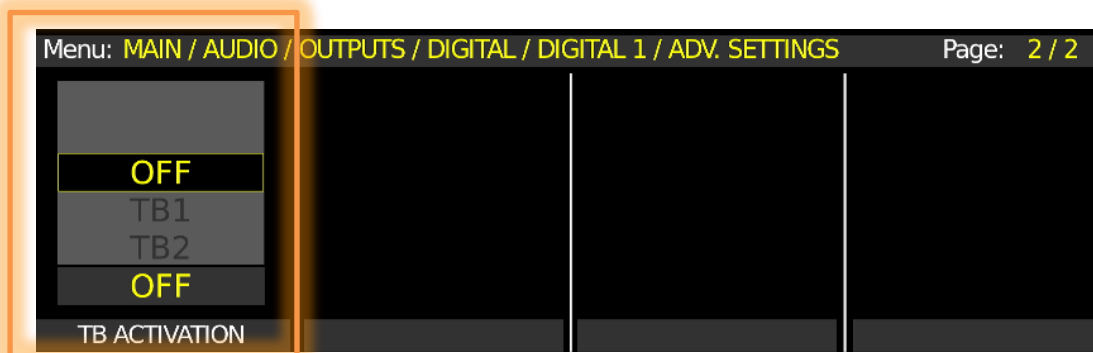


Determines whether the Control Room-to-Studio Talkback signal is included.

- **OFF** – Talkback audio is not sent to DIGITAL 1.

- **FROM CR TO ST** – Routes the Talkback signal generated in the Control Room and directed to the Studio.

TB ACTIVATION



Selects the command that activates the Talkback signal:

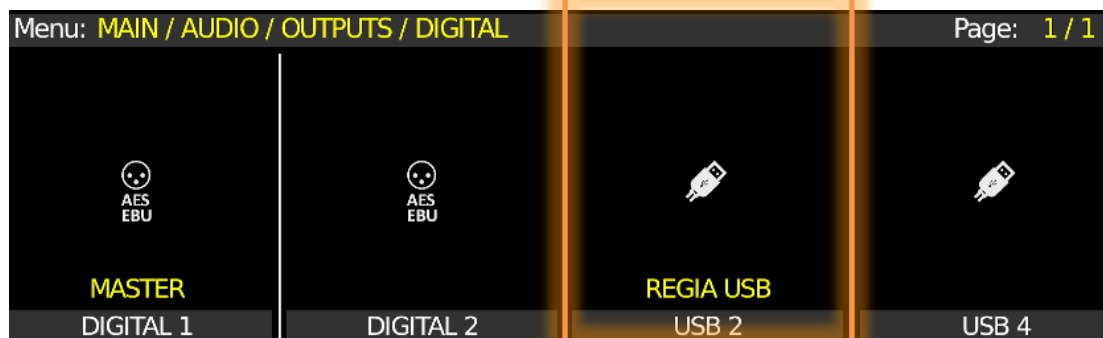
- **OFF** – Talkback activation is disabled.
- **TB1** – Activation by the TB1 command.
- **TB2** – Activation by the TB2 command.
- **TB3** – Activation by the TB3 command.
- **TB4** – Activation by the TB4 command.

TB SIGNAL selects the Talkback path, while **TB ACTIVATION** selects the command that enables it.

USB 2:

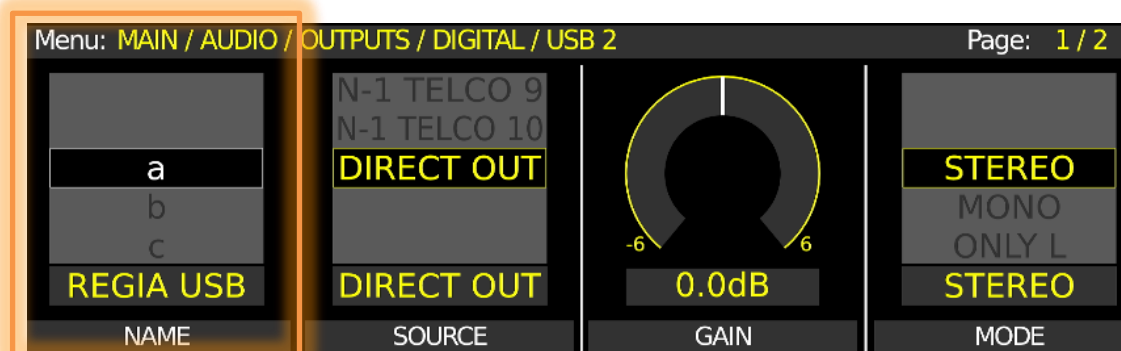
USB 2 is a bidirectional USB audio connection that allows the Oxygen 3000 Plus to exchange digital audio directly with a computer, without requiring an external sound card.

When the console is connected through a **USB Type-B 2.0 cable**, the computer detects it as a USB audio device. The USB 2 output sends audio from the mixer to recording, streaming or communication software installed on the computer.



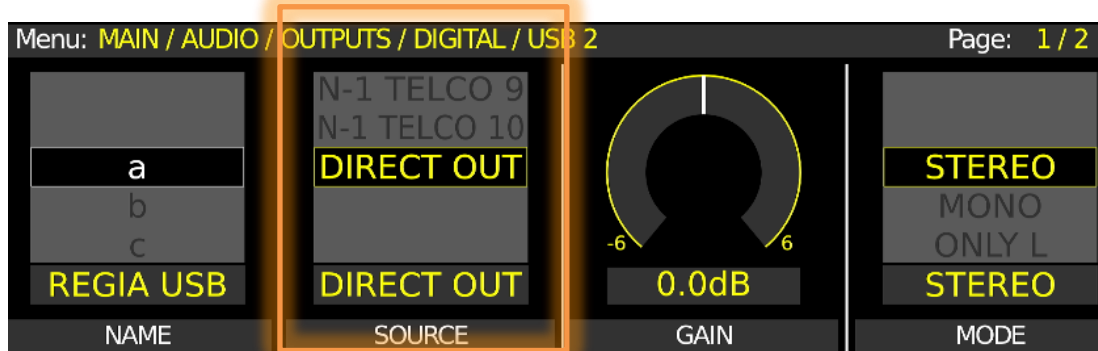
INPUTS SETTING (USB 2)

NAME:



Assigns a customised name to the USB output. The label can contain up to **8 characters**. Changing the name only affects identification within the console interface and **does not modify the audio signal**.

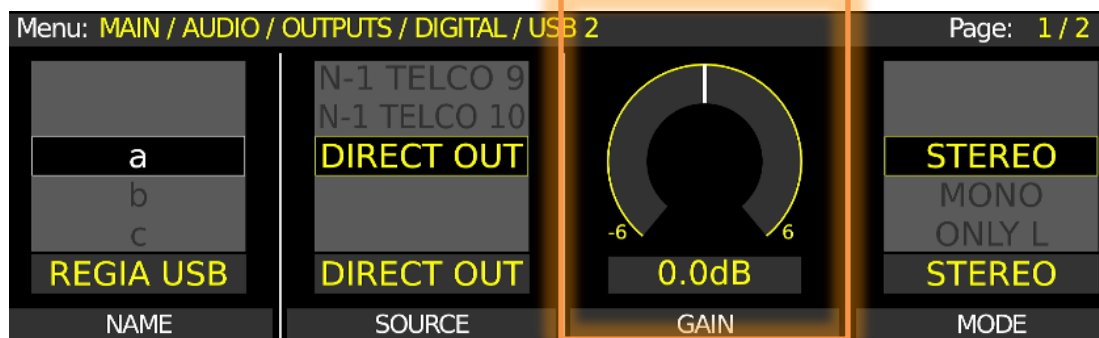
SOURCE:



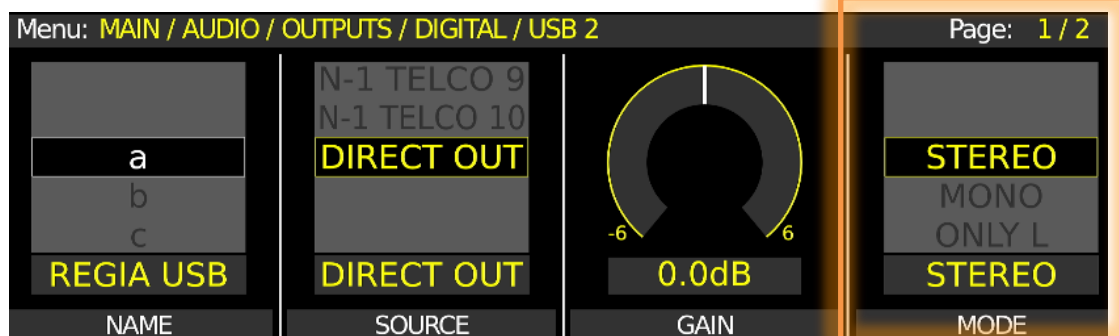
Selects the internal mixer signal sent to the computer through USB 2. The available options include:

- **PGM** – Sends the main Programme bus.
- **SUB** – Sends the secondary programme bus.
- **AUX 1** – Sends the first independent auxiliary mix.
- **AUX 2** – Sends the second independent auxiliary mix.
- **HDP CRM** – Sends the Control Room headphone-monitoring signal.
- **SPK CRM** – Sends the Control Room loudspeaker-monitoring signal.
- **HDP STD** – Sends the Studio headphone-monitoring signal.
- **SPK STD** – Sends the Studio loudspeaker-monitoring signal.
- **HDP CRM 90%** – Sends the 90% level variant of the Control Room headphone signal.
- **SPK CRM 90%** – Sends the 90% level variant of the Control Room loudspeaker signal.
- **HDP STD 90%** – Sends the 90% level variant of the Studio headphone signal.
- **SPK STD 90%** – Sends the 90% level variant of the Studio loudspeaker signal.
- **N-1 TELCO 3 to N-1 TELCO 10** – Sends the mix-minus return associated with the selected Telco channel.
- **DIRECT OUT:** Routes a selected input source directly to the Dante output without first using one of the main mixing buses.

The N-1 options remove the corresponding Telco input from its return feed, preventing the caller from receiving a delayed copy of their own voice. The **90% indication** identifies a firmware-defined level variant. It must not be interpreted as **-10 dB** or **90 dB**.

GAIN:

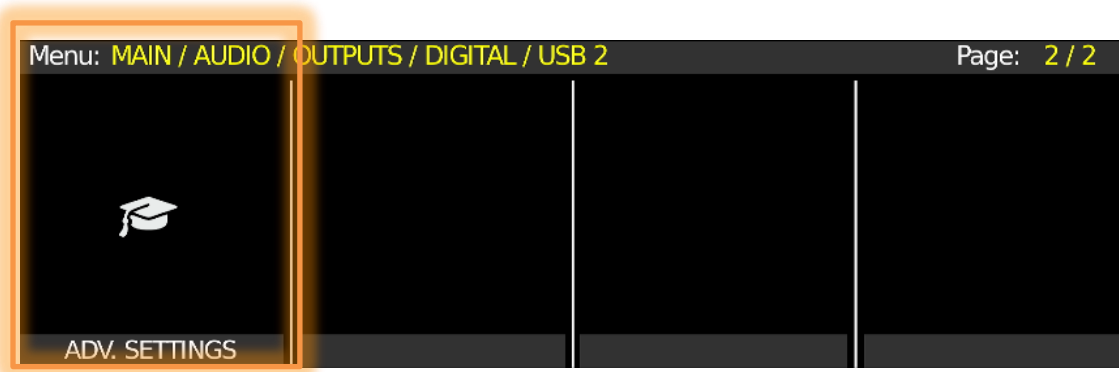
Adjusts the digital level sent from the console to the computer. **Range: -6.0 dB to +6.0 dB. Adjustment step: 0.1 dB.** At **0.0 dB**, the signal is sent at unity gain. Negative values attenuate the USB output, while positive values increase its digital level. The USB recording meter should be monitored carefully. Excessive positive gain may cause **digital clipping**, even if the computer's recording control is subsequently reduced.

MODE:

Determines how the Left and Right channels are sent to the computer.

- **STEREO** – Preserves the original independent Left and Right channels.
- **MONO** – Combines the Left and Right channels and sends the resulting mono mix on both USB channels.
- **MONO LEFT** – Uses only the Left channel and reproduces it on both USB channels.
- **MONO RIGHT** – Uses only the Right channel and reproduces it on both USB channels.

Although MONO modes are selected, the computer may continue to detect USB 2 as a **two-channel audio device**, with the same mono signal present on both channels.

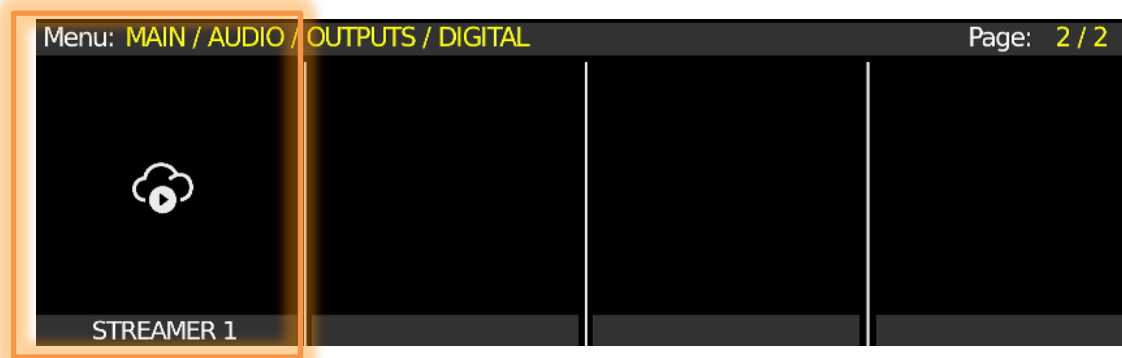
ADVANCED SETTINGS:

Provides additional routing, N-1 and Talkback controls for USB 2. The **ADVANCED SETTINGS** for **USB 2** are essentially identical to those previously described for **DIGITAL 1**. For further details beyond the information provided in the following sections, refer to the **ANALOG 1-2 (XLR)** section under **ANALOG OUTPUTS**.

STREAMER 1

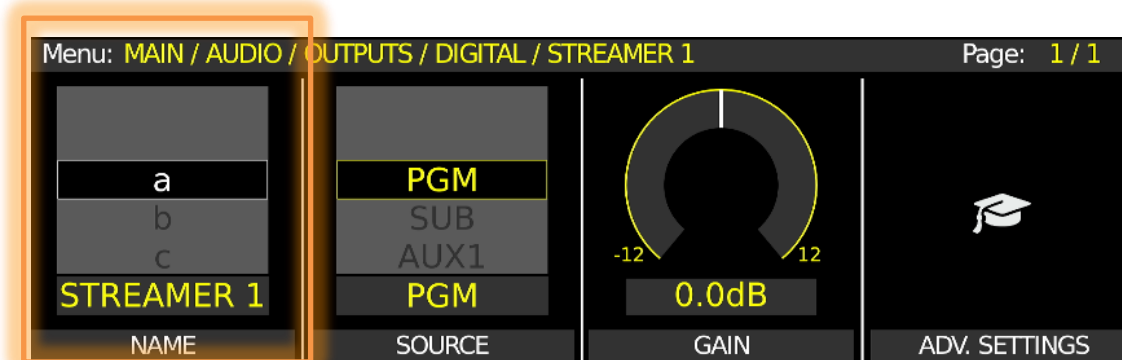
STREAMER 1 is the logical output that sends audio from the Oxygen 3000 Plus to its internal **streaming encoder**. The encoder converts the selected mixer signal into an IP audio stream that can be transmitted through the console's Ethernet connection to a local network, a remote receiver or an Internet streaming server. STREAMER 1 must not be confused with Dante: **Dante transports uncompressed professional audio over a dedicated AoIP network**, while STREAMER 1 uses the console's integrated encoder and standard IP streaming protocols.

The STREAMER 1 output page selects and prepares the audio sent to the encoder. Parameters such as **codec**, **bitrate**, **sample rate**, **destination address**, **port**, **protocol** and **server credentials** are configured separately through the Oxygen Web Interface.



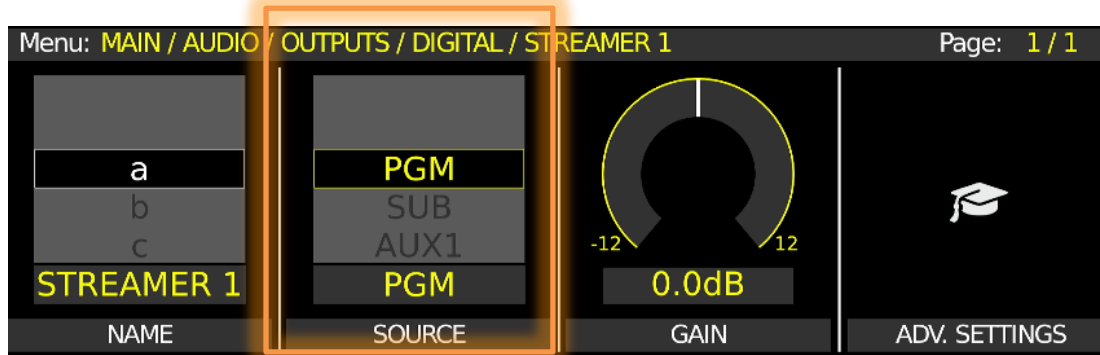
INPUTS SETTING (STREAMER 1)

NAME:



Assigns a customised name to the streaming output, making it easier to identify in the console interface. The label can contain up to **8 characters**. Changing the name **does not affect the encoded audio or network configuration**.

SOURCE:

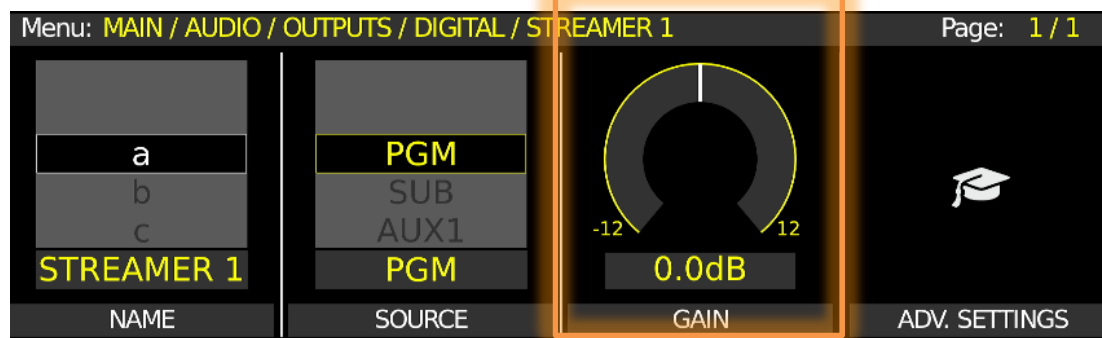


Selects the internal mixer signal sent to the streaming encoder. The available options include:

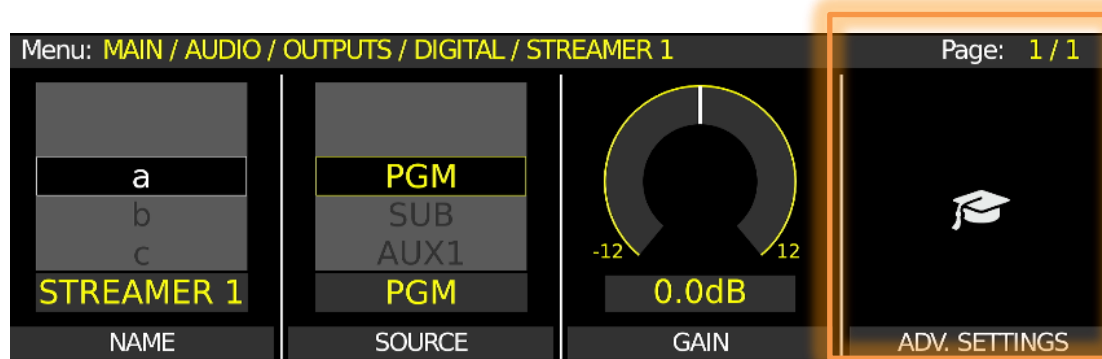
- **PGM** – Sends the main Programme bus.
- **SUB** – Sends the secondary programme bus.
- **AUX 1** – Sends the first independent auxiliary mix.
- **AUX 2** – Sends the second independent auxiliary mix.
- **HDP CRM** – Sends the Control Room headphone-monitoring signal.
- **SPK CRM** – Sends the Control Room loudspeaker-monitoring signal.
- **HDP STD** – Sends the Studio headphone-monitoring signal.
- **SPK STD** – Sends the Studio loudspeaker-monitoring signal.
- **HDP CRM 90%** – Sends the 90% level variant of the Control Room headphone signal.
- **SPK CRM 90%** – Sends the 90% level variant of the Control Room loudspeaker signal.
- **HDP STD 90%** – Sends the 90% level variant of the Studio headphone signal.
- **SPK STD 90%** – Sends the 90% level variant of the Studio loudspeaker signal.
- **N-1 TELCO 3 to N-1 TELCO 10** – Sends the mix-minus return associated with the selected Telco channel.
- **DIRECT OUT:** Routes a selected input source directly to the Dante output without first using one of the main mixing buses.

The N-1 options exclude the corresponding Telco input from its return feed, preventing the caller from receiving a delayed copy of their own voice.

SOURCE selects the audio sent to the encoder; it does not select the streaming destination, codec or network protocol.

GAIN:

Adjusts the digital audio level sent to the streaming encoder. **Range: -6.0 dB to +6.0 Db. Adjustment step: 0.1 dB.** Negative values attenuate the signal, while positive values increase it. The level should be set carefully because an overloaded signal may already be clipped before it reaches the encoder. The encoder input meter should remain below **0 dBFS**, leaving sufficient headroom for programme peaks.

ADV. SETTINGS:

Provides direct routing, N-1 and Talkback controls for the STREAMER 1 output. The **ADVANCED SETTINGS** for **STREAMER 1** are essentially identical to those previously described for **DIGITAL 1**. For further details beyond the information provided in the following sections, refer to the **ANALOG 1-2 (XLR)** section under **ANALOG OUTPUTS**.

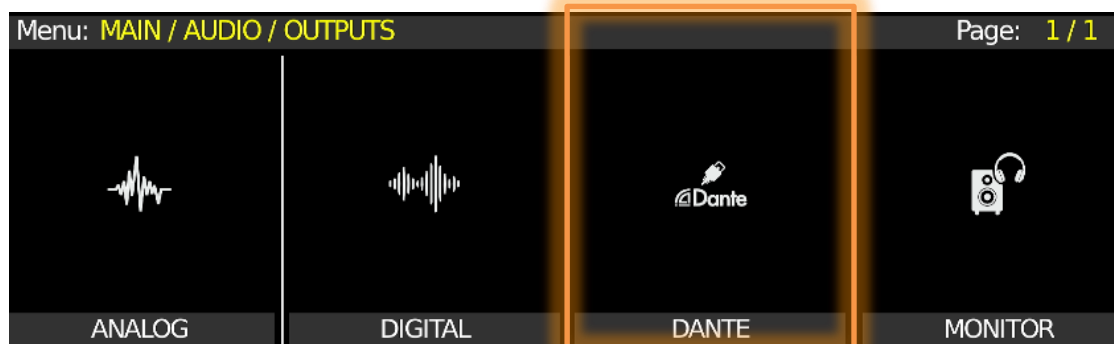
3.3 DANTE OUTPUT* (OPTIONAL)

The **DANTE OUTPUT** section manages the audio signals transmitted by the Oxygen 3000 Plus over a **Dante Audio-over-IP network**. The optional Dante interface provides multiple digital input and output channels through a dedicated Ethernet connection.

Dante outputs can be routed to compatible receivers—such as another console, an audio interface, an audio processor or a monitoring system—using Dante network-management software.

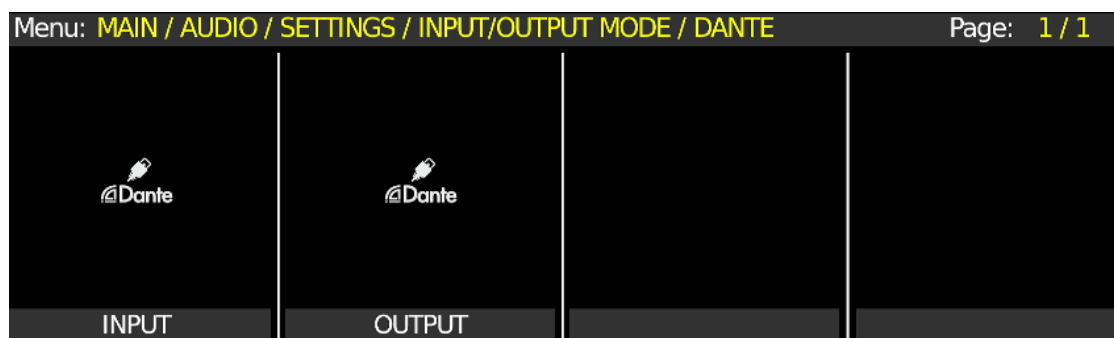
Two separate configurations must be considered:

- **INPUT/OUTPUT MODE** defines the structure and intended use of each Dante channel pair.
- **SOURCE** selects the internal Oxygen 3000 Plus signal transmitted through a specific Dante output.



DANTE INPUT/OUTPUT MODE:

Within **INPUT/OUTPUT MODE**, each Dante pair can be configured according to the type of signal that must be carried. It should be noted that, by default, all **DANTE OUTPUTS** are configured in **STEREO mode**.

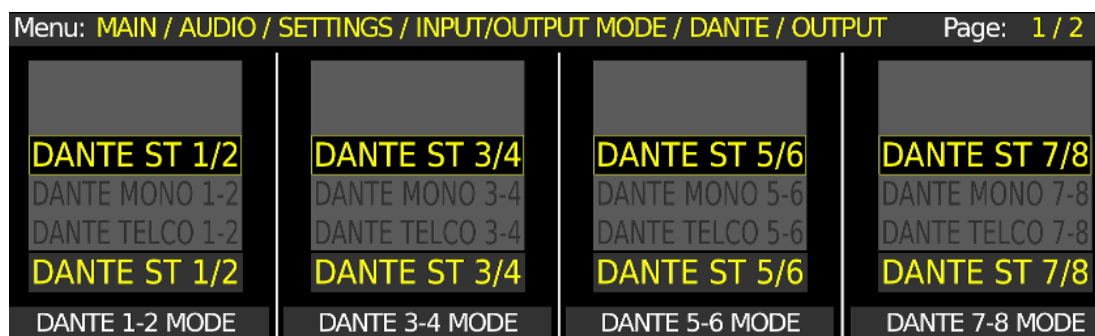


Selectable modes:

- **DANTE ST**: Configures the selected Dante pair as **one stereo channel**, preserving the independent left and right signals.
- **DANTE MONO**: Divides the Dante pair into **two independent mono channels**. Each side of the original pair can therefore be assigned and managed separately.
- **DANTE TELCO**: Divides the Dante pair into **two independent TELCO channels**, intended for external telephone hybrids or other bidirectional communication devices requiring **clean-feed/N-1 signals**.

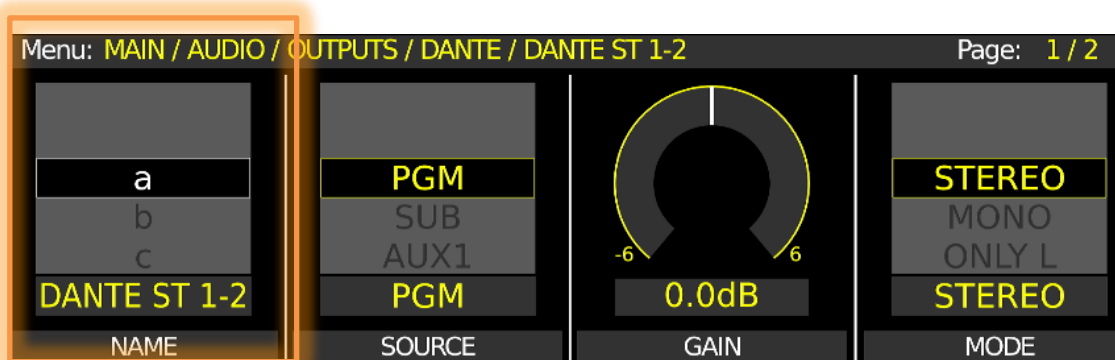
This configuration is applied independently to each available Dante pair. Consequently, different pairs can be used simultaneously for different purposes: one pair may carry a stereo programme, another may provide two mono signals and another may be configured for two TELCO connections.

Important: Changing a Dante pair from **DANTE ST** to **DANTE MONO** or **DANTE TELCO** changes the sources and outputs made available in the console menus. Existing assignments should therefore be checked after changing this setting.



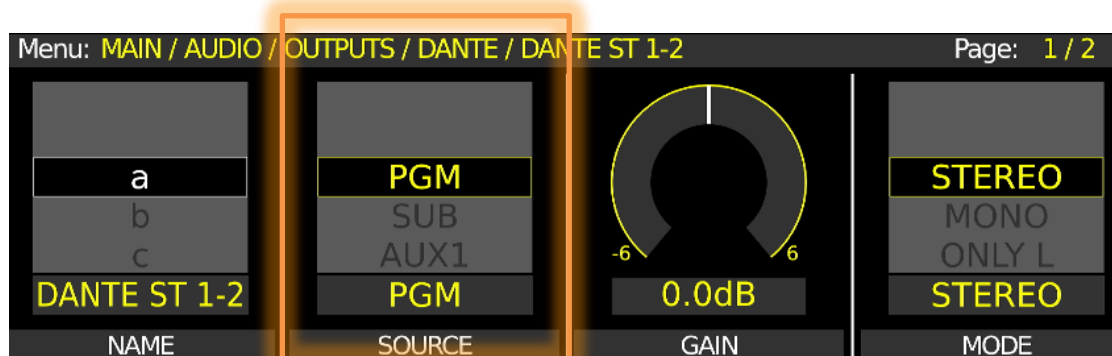
INPUTS SETTING (DANTE OUTPUT)

NAME:



Assigns a customised name to the selected Dante output, making it easier to identify within the Oxygen 3000 Plus configuration menu. The label can contain up to **8 characters**. The name identifies the output within the console. It does not, by itself, establish a Dante network connection or audio subscription.

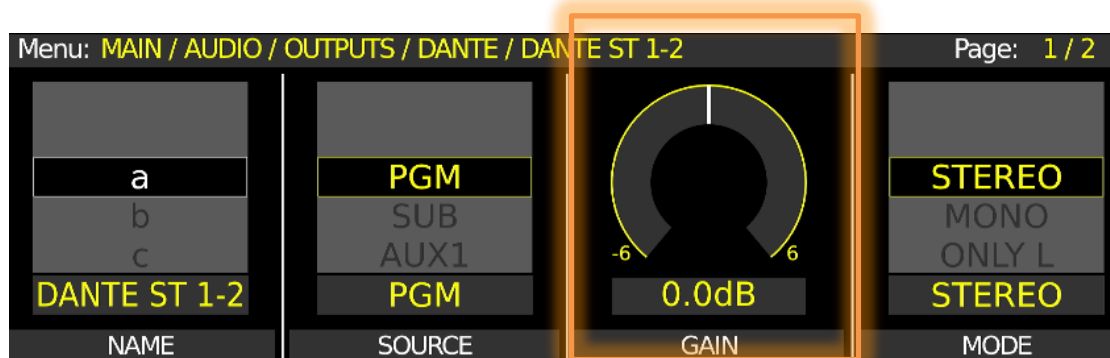
SOURCE:



The **SOURCE** parameter selects the internal console signal routed to the selected Dante output or Dante output pair. **Selectable options:**

- **PGM:** Routes the main Programme bus to the Dante output.
- **SUB:** Routes the Sub bus.
- **AUX 1:** Routes the Auxiliary 1 bus.
- **AUX 2:** Routes the Auxiliary 2 bus.
- **SPK-CR:** Routes the Control Room loudspeaker-monitoring signal.
- **HDP-CR:** Routes the Control Room headphone-monitoring signal.
- **SPK-STD:** Routes the Studio loudspeaker-monitoring signal.
- **HDP-STD:** Routes the Studio headphone-monitoring signal.
- **N-1 TELCO 3–10:** Routes the dedicated **clean-feed/N-1 signal** associated with the selected TELCO source, from **TELCO 3 to TELCO 10**.
- **DIRECT OUT:** Routes a selected input source directly to the Dante output without first using one of the main mixing buses.

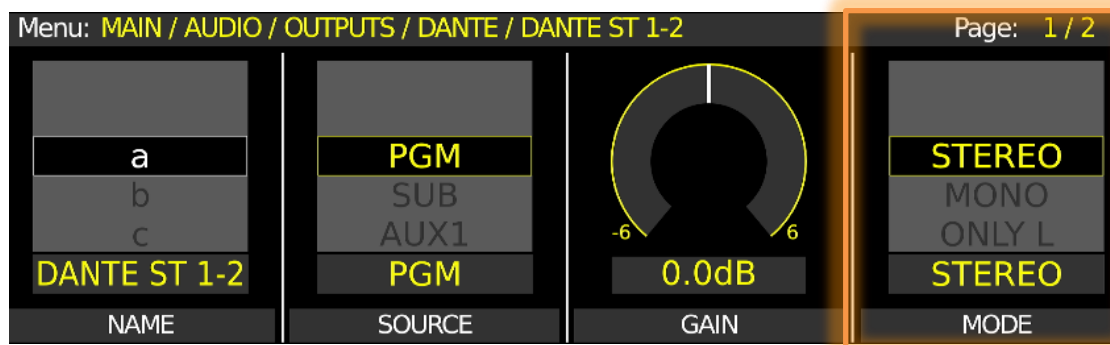
GAIN:



Adjusts the level of the selected Dante output before it is transmitted to the Dante network.

Range: **–6.0 dB to +6.0 dB. Adjustment Step: 0.1 dB.** Positive values increase the output level, while negative values attenuate it. **Important:** Excessive gain may cause digital clipping in the receiving Dante device. Monitor the output level before applying positive gain.

MODE:



Determines how the left and right channels of the selected source are processed at the Dante output.

Selectable options:

- **STEREO:** Preserves the original left and right channels as a stereo signal.
- **MONO:** Combines the stereo source into a mono signal.
- **MONO LEFT:** Uses the left channel of the selected source as the mono output signal.
- **MONO RIGHT:** Uses the right channel of the selected source as the mono output signal.

The **MODE** parameter must not be confused with **DANTE INPUT/OUTPUT MODE**:

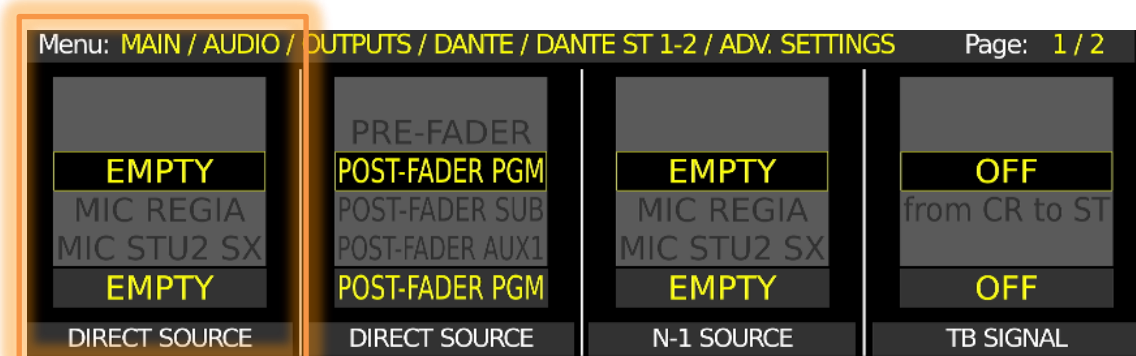
- **DANTE INPUT/OUTPUT MODE** defines whether the Dante pair operates as **stereo, two mono channels or two TELCO channels**.
- The output **MODE** parameter determines how the selected source's left and right components are handled at that particular output.

ADVANCED SETTINGS:

Opens the advanced routing and signal-management settings associated with the selected Dante output.



DIRECT SOURCE – Source Selection:



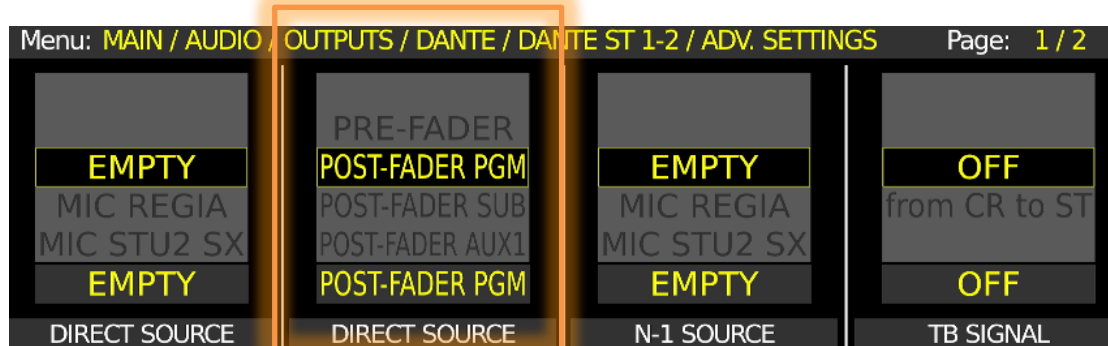
Selects the input source that is sent directly to the Dante output when **SOURCE** is set to **DIRECT OUT**.

Selectable options:

- **EMPTY:** No input source is assigned to the direct output.
- **All available Oxygen 3000 Plus input sources:** Any input currently enabled in the console configuration can be selected.

Depending on the installed hardware and the current **INPUT/OUTPUT MODE** configuration, the available sources may include microphones, mono or stereo inputs, digital inputs, USB, BT, streaming, telephone, TELCO and Dante sources. **Important:** The selectable source list changes according to the installed options and the current input-mode configuration.

DIRECT SOURCE – Signal Take-Off Point:



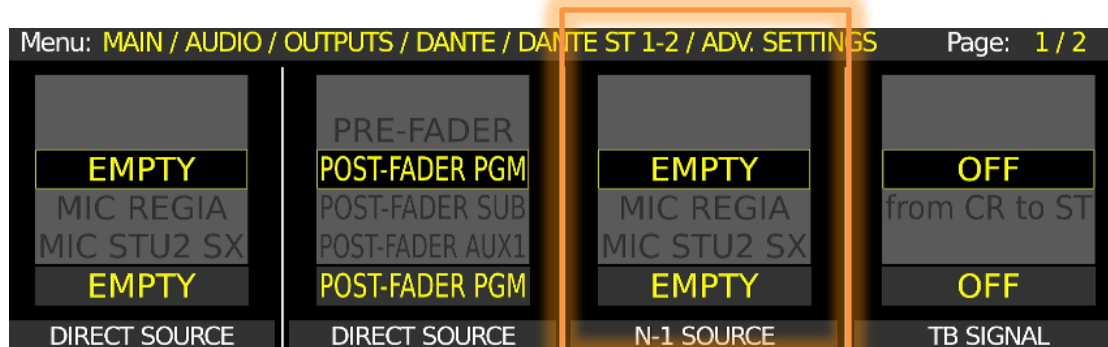
Determines where the selected direct signal is taken from within the channel's signal path.

Selectable options:

- **PRE FADER:** Takes the signal before the channel fader. Changes made to the fader level do not affect the Dante direct output.
- **POST FADER PGM:** Takes the channel's post-fader contribution to the **PGM** bus.
- **POST FADER SUB:** Takes the channel's post-fader contribution to the **SUB** bus.
- **POST FADER AUX 1:** Takes the channel's post-fader contribution to the **AUX 1** bus.
- **POST FADER AUX 2:** Takes the channel's post-fader contribution to the **AUX 2** bus.

The **PRE FADER** option is useful for recording or external processing when a constant source level is required. A **POST FADER** option should be selected when the Dante output must follow the channel level used for a specific bus. **Important:** A POST FADER signal follows changes made with the channel fader and may also depend on its assignment to the selected bus.

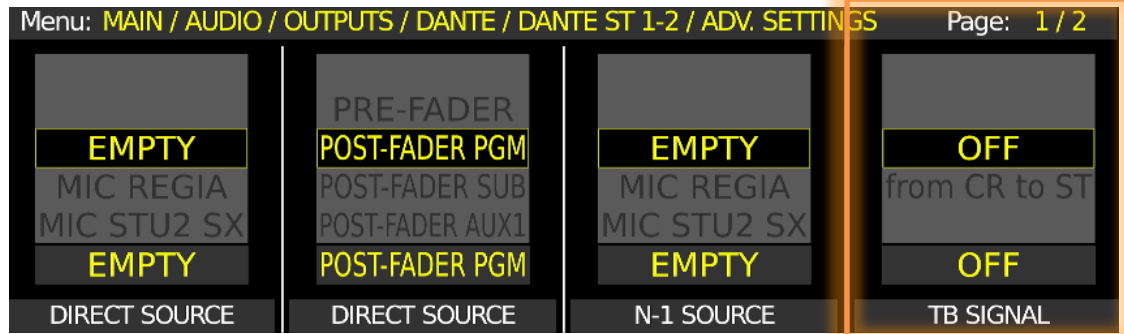
N-1 SOURCE:



Selects the input signal that must be excluded from the return mix when the Dante output is configured as an **N-1 TELCO** output. **Selectable options:**

- **All available Oxygen 3000 Plus input sources**

The selected source should normally correspond to the incoming audio from the TELCO device associated with that output. The console removes this signal from the return mix so that the caller does not hear their own voice. For example, when configuring **N-1 TELCO 5**, the incoming source associated with **TELCO 5** should be selected as the **N-1 SOURCE**. **Important:** Selecting the wrong N-1 source may cause the caller to hear their own delayed audio or may remove the wrong signal from the return feed.

TB SIGNAL:

Determines whether the Control Room-to-Studio Talkback signal is included in the selected Dante output.

Selectable options:

- **OFF:** No Talkback signal is sent to the Dante output.
- **FROM CR TO ST:** Routes the Talkback communication generated in the **Control Room** towards the **Studio** through the selected Dante output.

This function can be used when the Dante output feeds studio headphones, loudspeakers, a remote studio or another communication destination.

TB ACTIVATION:

Selects which dedicated Talkback button activates the Talkback signal on the selected Dante output.

Selectable options:

- **OFF:** No Talkback button is assigned.
- **TB1:** Talkback is activated by the **TB1** button.
- **TB2:** Talkback is activated by the **TB2** button.
- **TB3:** Talkback is activated by the **TB3** button.
- **TB4:** Talkback is activated by the **TB4** button.

The selected button controls when the signal defined under **TB SIGNAL** is sent to the output. **Important:** If **TB SIGNAL** is set to **OFF**, selecting TB1, TB2, TB3 or TB4 under **TB ACTIVATION** does not produce an audible Talkback signal on that output.

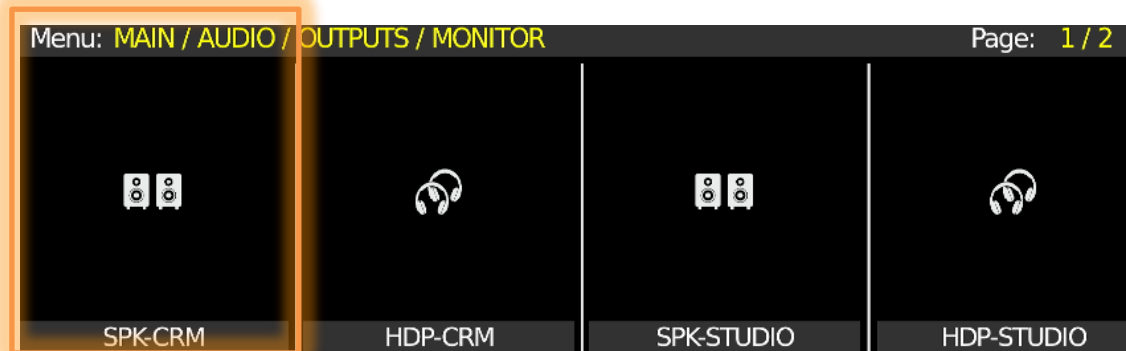
3.4 MONITOR OUTPUT

SPK-CRM:

The **SPK-CRM** section configures the loudspeaker-monitoring path used in the Control Room.

It can be accessed through: **MAIN → AUDIO → OUTPUTS → MONITOR → SPK-CRM**

The physical Control Room loudspeaker encoder is used to adjust the listening level. Pressing the encoder manually mutes or unmutes the loudspeakers. **Important:** SPK-CRM controls the Control Room monitoring path. It does not change the level of PGM, SUB, AUX-1, AUX-2 or the source being monitored.

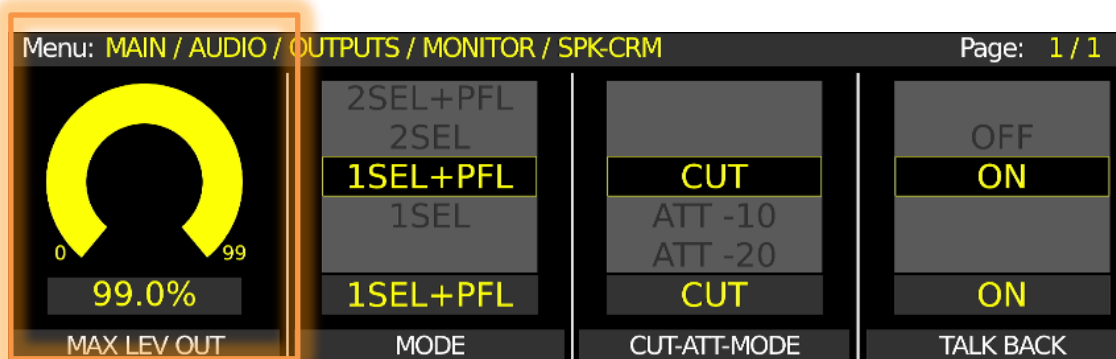


The SPK-CRM page includes:

- **MAX LEVEL OUT**
- **MODE**
- **CUT-ATT-MODE**
- **TALK BACK**

INPUTS SETTING (SPK-CRM)

MAX LEVEL OUT:



MAX LEVEL OUT limits the highest level that can be reached through the SPK-CRM loudspeaker-level control.

Range: 0.0 % to 99.0%. Adjustment step: 1.0%.

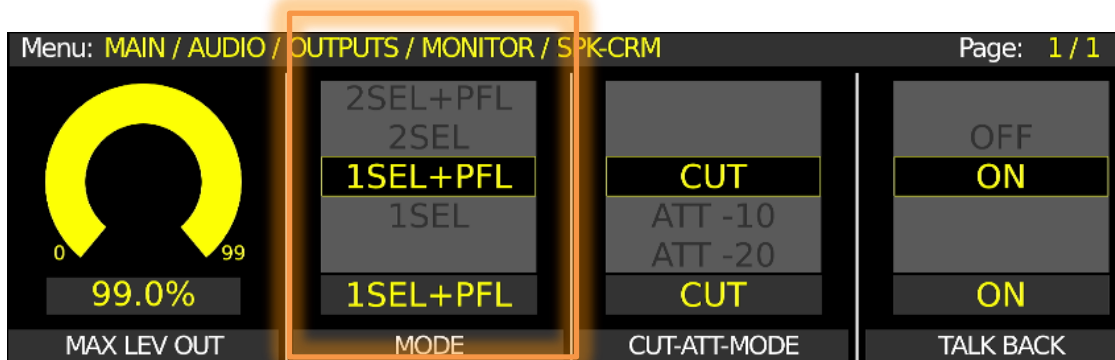
- **0%** sets the lowest permitted maximum level. This normally prevents a useful loudspeaker level from being obtained through the SPK-CRM control. **Important:** MAX LEVEL OUT at 0% is not the

same as the normal manual MUTE command. MUTE can be switched quickly from the surface, while MAX LEVEL OUT is a configuration limit.

- **Intermediate values** establish a reduced maximum loudspeaker level. The percentage does not necessarily represent an acoustic sound-pressure level or a direct dB value. It represents the permitted range of the console's monitor-level control.
- **99%** allows the complete normal SPK-CRM control range. This is the default setting.

Warning: MAX LEVEL OUT does not prevent clipping or loudspeaker damage caused by incorrect external amplifier settings. Calibrate the complete monitoring chain.

MODE:



MODE defines how sources are selected for the Control Room loudspeakers and whether channel PFL signals are reproduced through SPK-CRM. The selectable options are:

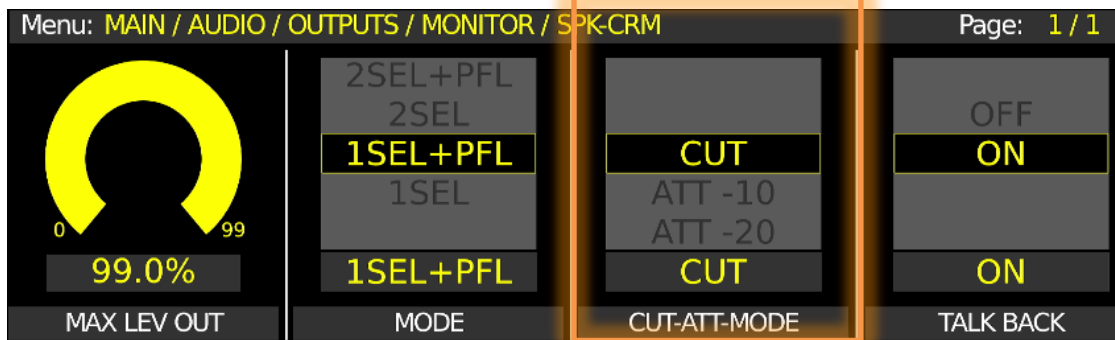
- **1SEL + PFL:** uses one normal monitor-source selection and also enables PFL reproduction through the Control Room loudspeakers. In this mode: One programme or monitor source is selected, The selected source is normally heard through SPK-CRM, Pressing a channel PFL button makes the PFL signal available to SPK-CRM, The operator can check a source before placing it On Air.
- **1SEL:** enables a single common monitor-source selection.
- **2SEL** enables dual-source monitoring without PFL.
- **2SEL+PFL:** enables dual-source monitoring with PFL support.

The **SEL + PFL** option allows channel PFL audio to reach the Control Room loudspeakers, while the other options directly select a fixed monitoring source.

CUT-ATT-MODE:

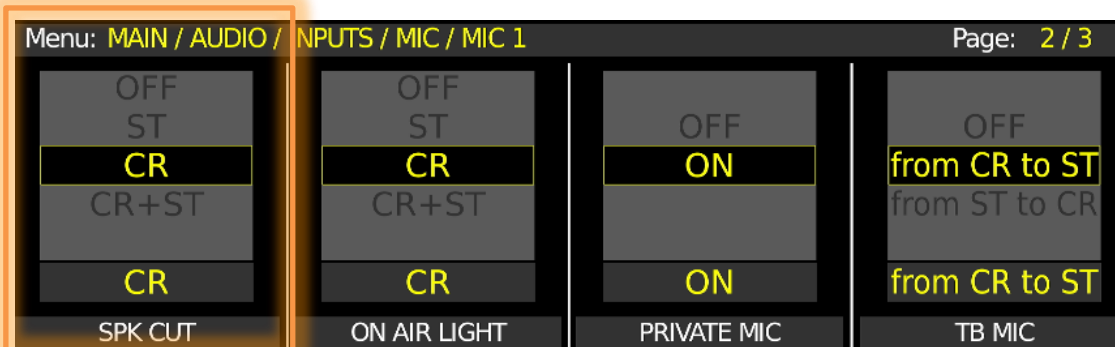
CUT-ATT-MODE determines what happens to the Control Room loudspeaker level when a microphone configured to affect the Control Room speakers becomes active. The selectable options are:

- **CUT:** completely mutes the SPK-CRM output when an assigned Control Room microphone becomes active. CUT provides the highest level of acoustic-feedback protection.
- **-40 dB:** strongly attenuates the Control Room loudspeakers while the assigned microphone is active.
- **-30 dB:** applies a substantial level reduction.
- **-20 dB:** applies a moderate-to-strong attenuation.
- **-10 dB:** applies the smallest available attenuation.



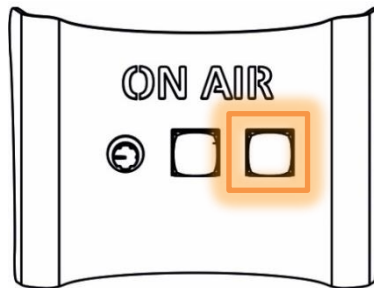
The trigger is generated when a microphone whose **SPK-CUT** assignment includes **CR** changes to its active state. The relevant microphone parameter can normally be found through:

MAIN → AUDIO → INPUTS → MIC → MIC 1–10 → SPK-CUT

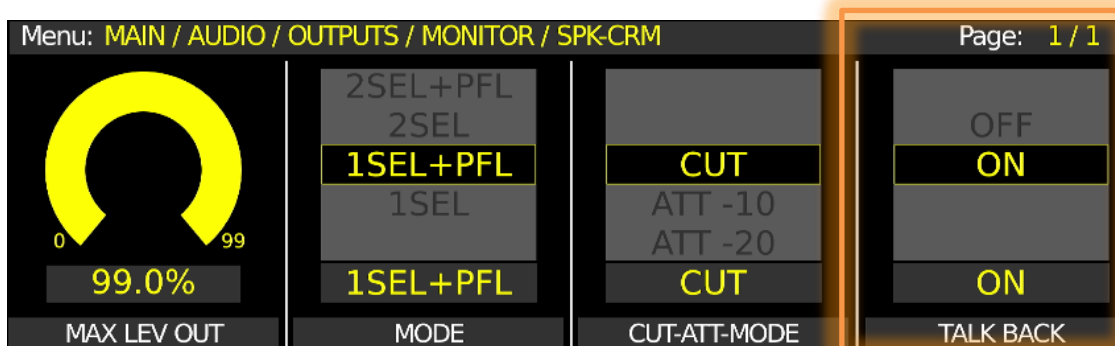


TALK BACK:

TALK BACK determines whether Talkback communication is reproduced through the Control Room loudspeakers.



- **OFF:** No Talkback communication from the studio is reproduced through the Control Room loudspeakers.
- **ON:** Talkback communication from the studio directed towards the Control Room is reproduced through SPK-CRM.

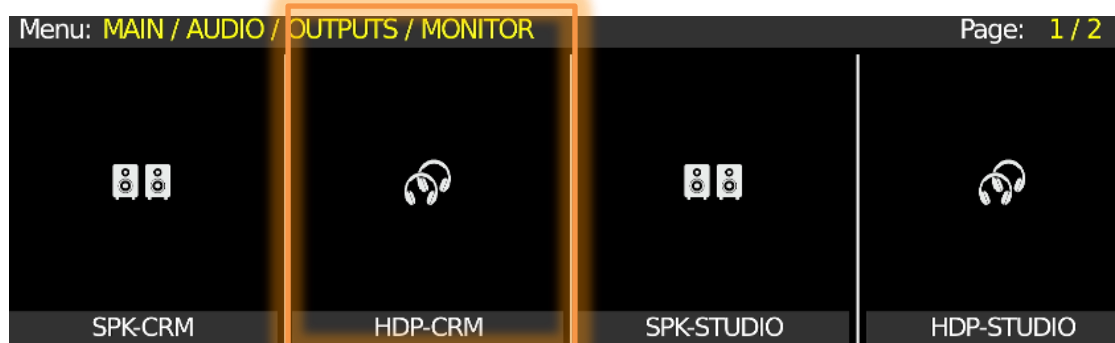


HDP-CRM:

The **HDP-CRM** section configures the headphone-monitoring path used in the Control Room.

It can be accessed through: **MAIN → AUDIO → OUTPUTS → MONITOR → HDP-CRM**

Important: HDP-CRM controls only the Control Room headphone-monitoring path. It does not change the level of PGM, SUB, AUX-1, AUX-2 or the source being monitored.



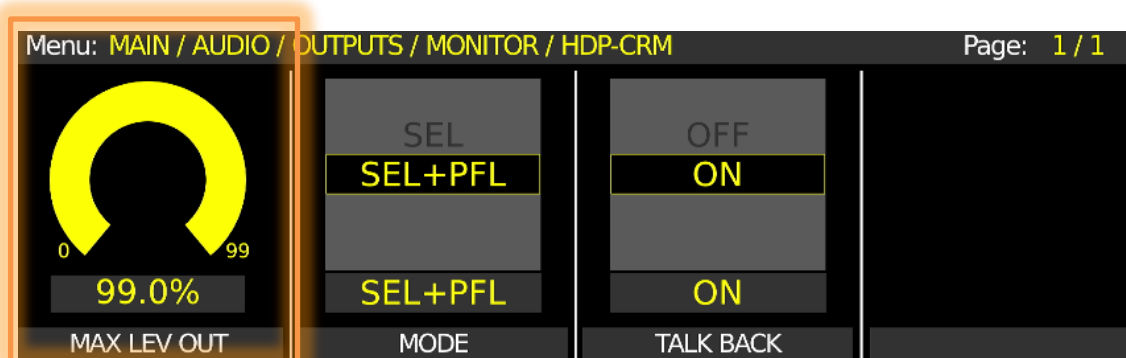
The HDP-CRM page includes:

- **MAX LEVEL OUT**
- **MODE**
- **TALK BACK**

The physical Control Room headphone encoder adjusts the current listening level within the limit established by MAX LEVEL OUT.

INPUTS SETTING (HDP-CRM)

MAX LEVEL OUT:



MAX LEVEL OUT limits the highest level that can be reached through the HDP-CRM headphone-level control.
Range: 0.0 % to 99.0%. Adjustment step: 1.0%.

- **0%** sets the lowest permitted maximum headphone level. This normally prevents a useful listening level from being obtained through HDP-CRM. **Important:** setting MAX LEVEL OUT to 0% is a configuration limit. It should not be confused with a temporary MUTE command.
- Intermediate values establish a reduced maximum headphone level. **Configuration note:** the percentage does not represent a direct acoustic sound-pressure level or a linear loudness

percentage. The actual result depends on headphone impedance, sensitivity and external amplification.

- **99%** makes the complete normal HDP-CRM level range available. This is the default value.

Important: MAX LEVEL OUT sets an upper limit. It does not establish the normal operating volume, which remains controlled by the HDP-CRM level encoder.

MODE:



MODE determines which source is reproduced through the Control Room headphones and whether channel PFL signals can be heard. The selectable options are:

- **SEL + PFL:** makes HDP-CRM follow the selected monitor source and also reproduces active channel PFL signals. **Important:** PFL is taken before the channel fader. The headphones may therefore reproduce a source even when its fader is completely closed.
- **SEL:** makes HDP-CRM follow the monitor source selected from the console surface. **Important:** SEL does not disable the PFL function on the console. It prevents PFL from being reproduced through this particular headphone path.

TALK BACK:



TALK BACK determines whether Talkback communication is reproduced through the Control Room headphones. For HDP-CRM, the Talkback signal normally travels from the Studio or another configured Talkback source towards the Control Room operator.

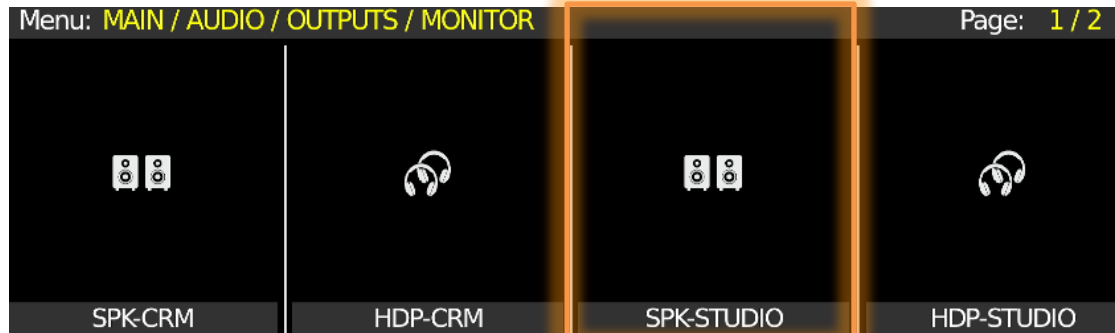
- **OFF:** No Talkback communication is reproduced through HDP-CRM.
- **ON:** Talkback communication directed towards the Control Room is reproduced through HDP-CRM.

SPK-STUDIO:

The **SPK-STUDIO** section configures the loudspeaker-monitoring path used inside the Studio.

It can be accessed through: **MAIN → AUDIO → OUTPUTS → MONITOR → SPK-STUDIO**

The physical Studio loudspeaker encoder is used to adjust the listening level. Pressing the encoder normally mutes or unmutes the Studio loudspeakers.

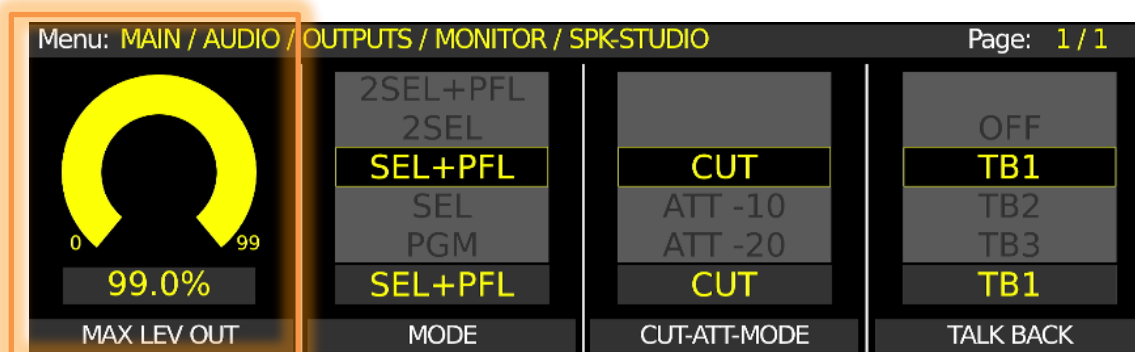


The SPK-STUDIO page includes:

- **MAX LEVEL OUT**
- **MODE**
- **CUT-ATT-MODE**
- **TALK BACK**

INPUTS SETTING (SPK-STUDIO)

MAX LEVEL OUT:

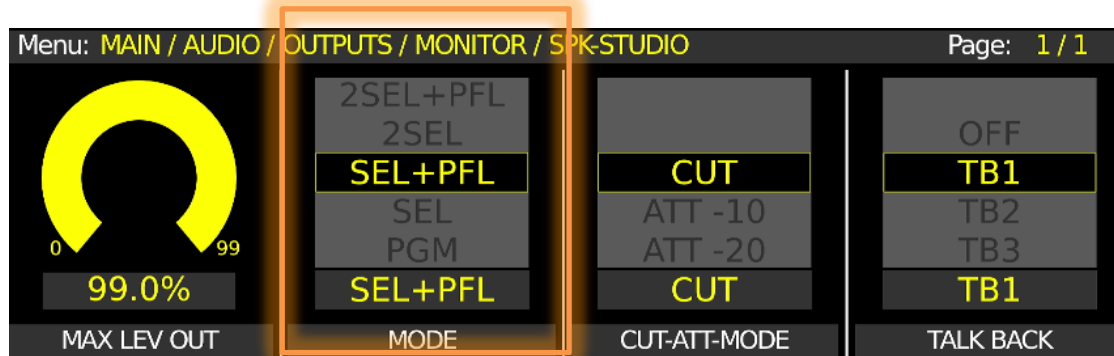


MAX LEVEL OUT limits the highest level that can be reached through the SPK-STUDIO loudspeaker-level control. **Range: 0.0 % to 99.0%. Adjustment step: 1.0%.**

- **0%** establishes the lowest possible maximum level. This normally prevents a useful loudspeaker level from being obtained through the SPK-STUDIO control. **Important:** setting MAX LEVEL OUT to 0% is different from manually muting the loudspeakers. Manual MUTE is an operating control, while MAX LEVEL OUT is a stored configuration limit.
- Intermediate values establish a reduced maximum Studio loudspeaker level. **Configuration note:** a 50% setting does not necessarily produce half the perceived loudness of a 99% setting. The result also depends on amplifier gain, loudspeaker sensitivity, room acoustics and listening distance.

- **99%** allows the complete normal SPK-STUDIO control range. This is the default value.

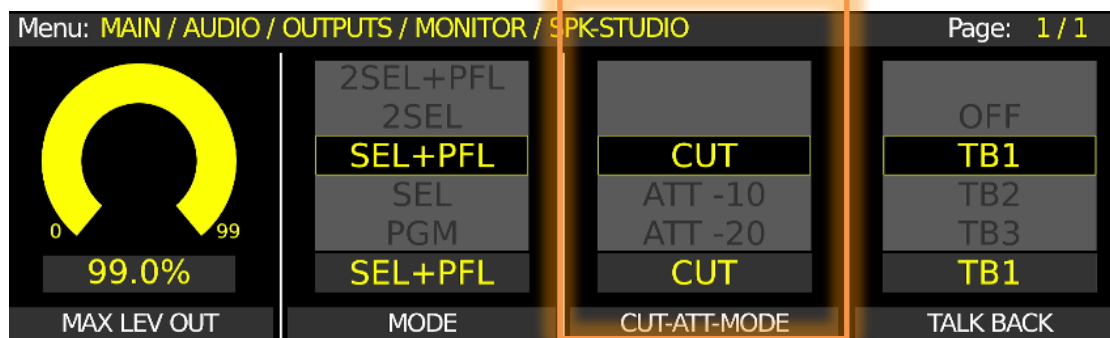
MODE:



MODE determines which signal is reproduced through the Studio loudspeakers and whether channel PFL signals can be heard through SPK-STUDIO. The selectable options are:

- **2SEL + PFL:** Provides the two-selection workflow and also allows active channel PFL signals to be reproduced through SPK-STUDIO.
- **2SEL:** Provides two separate monitor-source selections. The source associated with the Studio speaker control can be selected independently according to the surface monitoring logic.
- **SEL + PFL:** uses the normal monitor-source selection and also allows channel PFL signals to be reproduced through SPK-STUDIO.
- **SEL:** enables a common selectable monitor source.
- **PGM:** **PGM** directly assigns the Studio loudspeakers to the main programme bus.
- **SUB:** directly assigns the Studio loudspeakers to the SUB bus.
- **AUX1:** assigns the Studio loudspeakers to AUX1.
- **AUX2:** assigns the Studio loudspeakers to AUX2.
- **EXT1:** assigns the Studio loudspeakers to external monitor input 1.
- **EXT2:** assigns the Studio loudspeakers to external monitor input 2.

CUT-ATT:



CUT-ATT determines what happens to the Studio loudspeaker level when a microphone configured to affect the Studio speakers becomes active. The selectable options are:

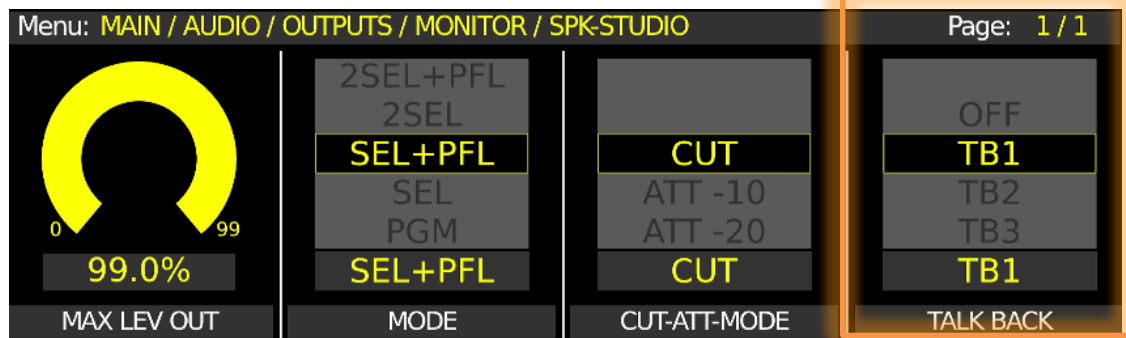
- **CUT**: completely mutes SPK-STUDIO when an assigned Studio microphone becomes active. CUT provides the highest protection against acoustic feedback.
- **-40 dB**: applies very strong attenuation while the assigned microphone is active. The monitoring source
- **-30 dB**: applies a substantial level reduction.
- **-20 dB**: applies a moderate-to-strong attenuation.
- **-10 dB**: applies the smallest available attenuation.

The trigger is generated when a microphone whose **SPK-CUT** assignment includes the Studio changes to its active state. The corresponding microphone parameter can normally be found through:

MAIN → AUDIO → INPUTS → MIC → MIC 1–10 → SPK-CUT



TALK BACK:



TALK BACK determines whether Talkback communication is reproduced through the Control Room headphones. The selectable options are:

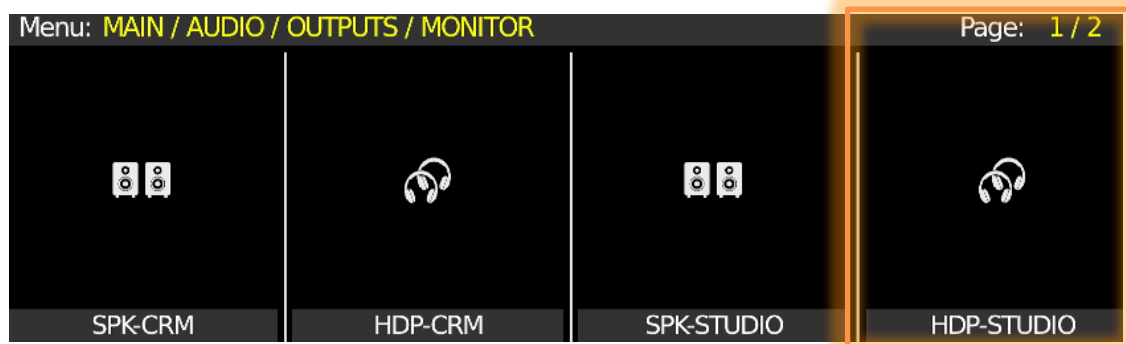
- **OFF:** No Talkback bus is reproduced through HDP-CRM.
- **TB1:** The Talkback signal associated with **TB1** is reproduced through the Control Room headphones when TB1 becomes active.
- **TB2:** The Talkback signal associated with **TB2** is reproduced through HDP-CRM when TB2 becomes active.
- **TB3:** The Talkback signal associated with **TB3** is reproduced through HDP-CRM when TB3 becomes active.
- **TB4:** The Talkback signal associated with **TB4** is reproduced through HDP-CRM when TB4 becomes active.

HDP-STUDIO:

The **HDP-STUDIO** section configures the headphone-monitoring path used inside the Studio.

It can be accessed through: **MAIN → AUDIO → OUTPUTS → MONITOR → HDP-STUDIO**

The physical Studio headphone encoder is used to adjust the listening level. Pressing the encoder normally mutes or unmutes the HDP-STUDIO output.



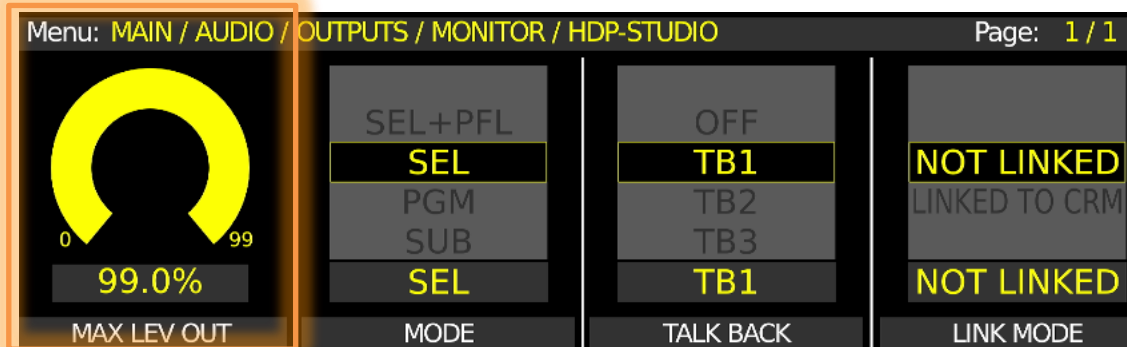
The HDP-STUDIO page includes:

- **MAX LEVEL OUT**
- **MODE**
- **TALK BACK**

- LINK MODE

INPUTS SETTING (HDP-STUDIO)

MAX LEVEL OUT:



MAX LEVEL OUT limits the highest level that can be reached through the HDP-STUDIO headphone-level control. **Range: 0.0 % to 99.0%. Adjustment step: 1.0%.**

- **0%** establishes the lowest possible maximum level. This normally prevents a useful headphone level from being obtained through the HDP-STUDIO control. **Important:** MAX LEVEL OUT at 0% is different from manual MUTE. MUTE is an operating control, while MAX LEVEL OUT is a stored configuration limit.
- Intermediate values establish a reduced maximum Studio headphone level. **Configuration note:** two headphone models connected at the same percentage may produce significantly different listening levels.
- **99%** allows the complete normal HDP-STUDIO control range. This is the default value.

Warning: headphone monitoring can produce high sound-pressure levels without being audible to the Control Room operator. Calibrate MAX LEVEL OUT before allowing another person to use the headphones.

MODE:



HDP-STUDIO is a monitoring path and not a programme bus. The selected **MODE** determines the audio source reproduced through the Studio headphones. The available options are:

- **SEL + PFL:** When MODE is set to **SEL+PFL**, HDP-STUDIO monitors the selected source and also allows PFL audio to be heard according to the console's PFL logic. SEL+PFL is suitable when the Studio listener must hear the normal selected source but also needs access to channel or bus PFL.
- **SEL:** When MODE is set to **SEL**, HDP-STUDIO monitors the source selected by the console's monitoring selection. SEL is suitable when the Studio should follow the operator's selected monitoring source without being permanently assigned to one programme bus.
- **PGM:** When MODE is set to **PGM**, HDP-STUDIO monitors the **PGM** bus. PGM is suitable for presenters and guests who must continuously hear the programme output.
- **SUB:** When MODE is set to **SUB**, HDP-STUDIO monitors the **SUB** bus. SUB is suitable when the Studio requires a secondary programme feed.
- **AUX1:** When MODE is set to **AUX 1**, HDP-STUDIO monitors the **AUX 1** bus. The content heard through HDP-STUDIO depends on the sources assigned to AUX 1 and their individual levels.
- **AUX2:** When MODE is set to **AUX 2**, HDP-STUDIO monitors the **AUX 2** bus. The content heard through HDP-STUDIO depends on the sources assigned to AUX 2 and their individual levels.
- **EXT1:** When MODE is set to **EXT 1**, HDP-STUDIO monitors the **EXT 1** input or external monitoring source. The exact content depends on the external routing configuration.
- **EXT2:** When MODE is set to **EXT 2**, HDP-STUDIO monitors the **EXT 2** input or external monitoring source. The exact content depends on the external routing configuration.

ATTENTION: When **LINK MODE** is set to **LINKED TO CRM**, the HDP-STUDIO source follows HDP-CRM and MODE no longer determines the active source in the normal way.

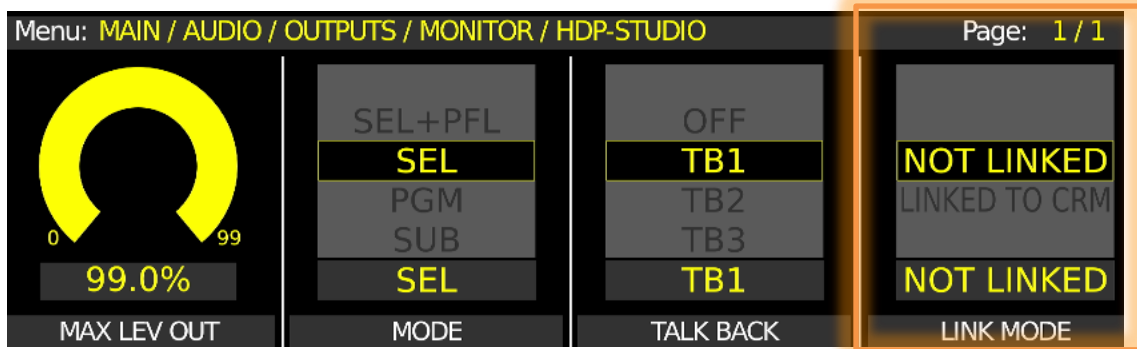
TALK BACK:



TALK BACK determines which Talkback communication is reproduced through the Studio headphones. The Talkback signal can remain available through HDP-STUDIO while the Studio loudspeakers are muted because a microphone is active. **Important:** TALK BACK selects the communication received by HDP-STUDIO. It does not configure the Talkback microphone, the communication direction or the Talkback activation button.

- **OFF:** No Talkback communication is reproduced through HDP-STUDIO.
- **TB1:** Only Talkback communication directed to **TB1** is reproduced through HDP-STUDIO.
- **TB2:** Only Talkback communication directed to **TB2** is reproduced through HDP-STUDIO.
- **TB3:** Only Talkback communication directed to **TB3** is reproduced through HDP-STUDIO.
- **TB4:** Only Talkback communication directed to **TB4** is reproduced through HDP-STUDIO.

LINK MODE:



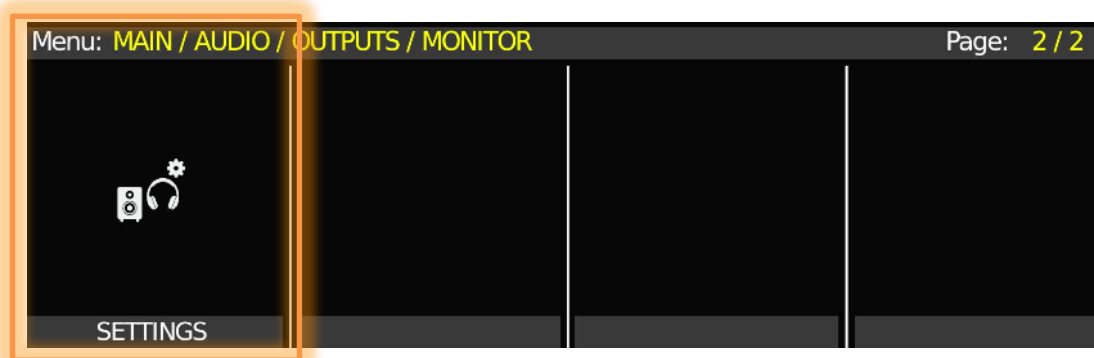
LINK MODE determines whether HDP-STUDIO uses an independent Studio monitoring source or follows the audio sent to the Control Room headphones. The selectable options are:

- **NOT LINKED:** When LINK MODE is set to **NOT LINKED**, HDP-STUDIO operates as an independent Studio headphone-monitoring path. NOT LINKED is **suitable** when: The presenter must continuously hear PGM - The Control Room operator frequently uses PFL - Studio personnel require a dedicated auxiliary mix.
- **LINKED TO CRM:** When LINK MODE is set to **LINKED TO CRM**, HDP-STUDIO follows the audio sent to HDP-CRM. LINKED TO CRM may be used when: The Control Room and Studio must hear the same signal - A producer and presenter must monitor identical programme material - A shared PFL or source-selection workflow is required. **Warning:** linked operation can make Control Room PFL or private material audible inside the Studio. Verify the complete workflow before selecting LINKED TO CRM during live operation.

SETTINGS:

The **SETTINGS** section configures the global functions shared by the Control Room and Studio monitoring paths. It can be accessed through: **MAIN → AUDIO → OUTPUTS → MONITOR → SETTINGS**

Unlike **SPK-CRM**, **HDP-CRM**, **SPK-STUDIO** and **HDP-STUDIO**, this page does not configure the level or MODE of one individual monitor output.

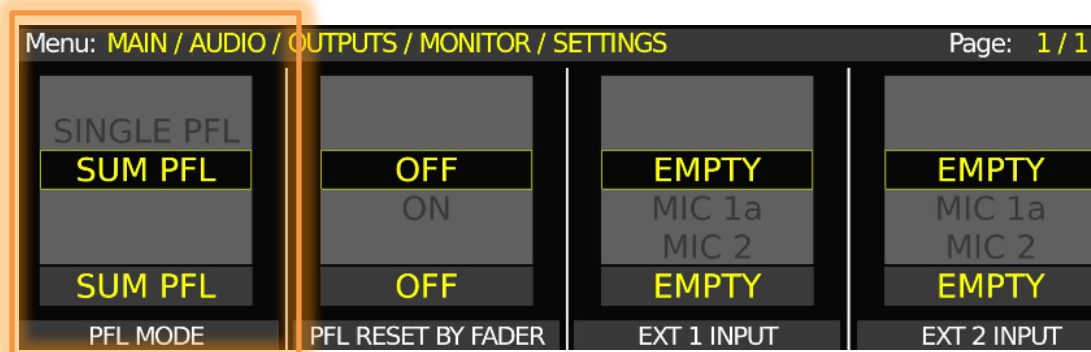


The SETTINGS page includes:

- **PFL MODE**
- **PFL RESET BY FADER**
- **EXT 1 INPUT**
- **EXT 2 INPUT**

INPUTS SETTING (SETTINGS)

PFL MODE:



PFL MODE determines how the console manages PFL when PFL is activated on more than one channel.

The selectable options are:

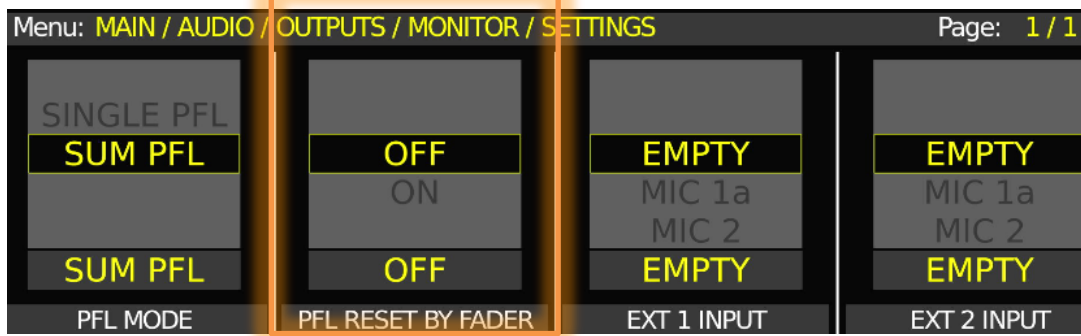
- **SINGLE PFL:** When **PFL MODE** is set to **SINGLE PFL**, only one channel PFL can remain active at a time. **In this mode:** Activating PFL on one channel selects that channel for pre-fader monitoring - Activating PFL on another channel automatically disables the previous PFL - The most recently selected PFL becomes the active monitoring source - The operator can move quickly from one source to another. **SINGLE PFL is suitable** when: Only one source must be checked at a time - Several simultaneous sources would make level adjustment difficult - Telephone or private communication must not be mixed with another PFL source - A simple pre-listening workflow is preferred.
- **SUM PFL:** When **PFL MODE** is set to **SUM PFL**, several channel PFL selections can remain active simultaneously. **In this mode:** Activating a new PFL does not automatically disable the previous PFL - Every active PFL source is added to the PFL bus - Multiple channels can be monitored together - Each PFL must be disabled manually or through PFL RESET BY FADER. **SUM PFL is suitable** when: Several microphones must be checked together - A group of sources must be monitored before transmission - Phase, balance or interaction between sources must be verified - A production workflow requires a temporary pre-fader mix - Several contribution signals must be compared simultaneously.,

PFL means **Pre-Fader Listen**. It allows a source to be monitored before the channel fader affects its level.

Important: PFL MODE does not activate PFL and does not determine which monitor output reproduces it.

PFL must be activated from a channel, and the required monitor output must use a MODE that includes PFL or follow another PFL-enabled monitor path.

PFL RESET BY FADER:

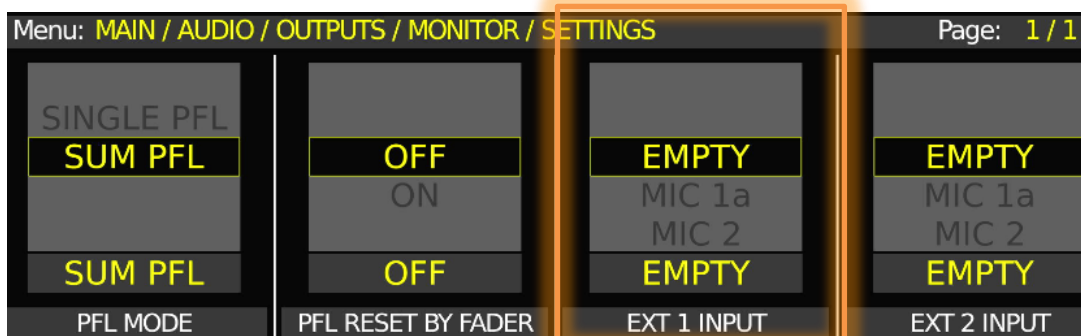


PFL RESET BY FADER determines whether the PFL of a channel is automatically disabled when its fader is opened. The selectable options are:

- **OFF:** When **PFL RESET BY FADER** is set to **OFF**, moving a channel fader does not automatically disable its PFL. OFF may be used when: Continuous PFL monitoring is required while the channel is active - The operator wants complete manual control of PFL - A production or technical test requires PFL to remain selected
- **ON:** When **PFL RESET BY FADER** is set to **ON**, opening a channel fader automatically disables the PFL associated with that channel. ON is suitable when: PFL is used only to prepare a source before transmission - The operator wants the normal monitor source to return automatically - Accidental continued PFL monitoring must be avoided - A fast live-broadcast workflow is required.

This function helps prevent a channel from remaining in pre-fader monitoring after the operator begins using it in the active mix. The reset is normally associated with the point at which the fader crosses the configured **FADER THRESHOLD** and the channel enters its active operating area. **Important:** PFL RESET BY FADER disables the PFL monitoring command. It does not mute the channel, move the fader or remove the channel from PGM, SUB, AUX 1 or AUX 2.

EXT 1 INPUT:



EXT 1 INPUT assigns an input source to the first external monitoring selection. When a monitor output is configured to reproduce **EXT 1**, it hears the source assigned under EXT 1 INPUT. The EXT 1 INPUT menu displays the input sources that can be used as external-monitor feeds on the installed console. Depending on hardware, INPUT MODE and firmware, the list can include:

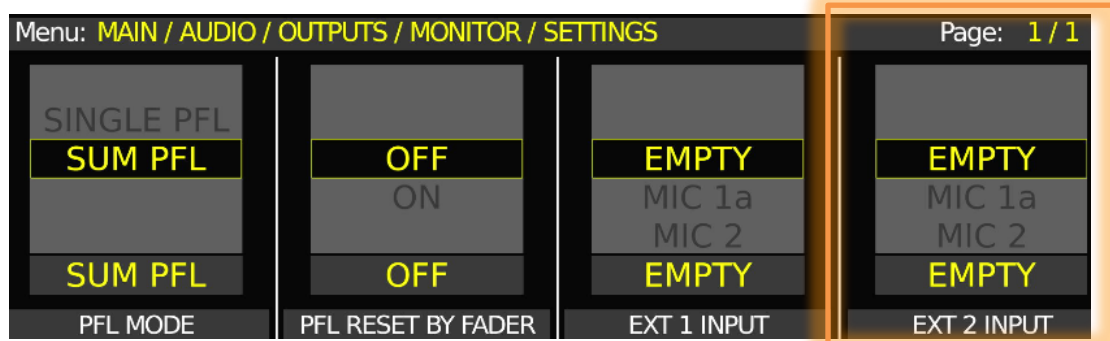
- Active **MONO** line-input sources
- Active **STEREO** line-input sources

- Corresponding **ANALOG** input labels
- Additional digital or network sources, when exposed by the installed firmware
- **NONE**, **EMPTY** or an equivalent unassigned state, when available

EXT 2 INPUT:

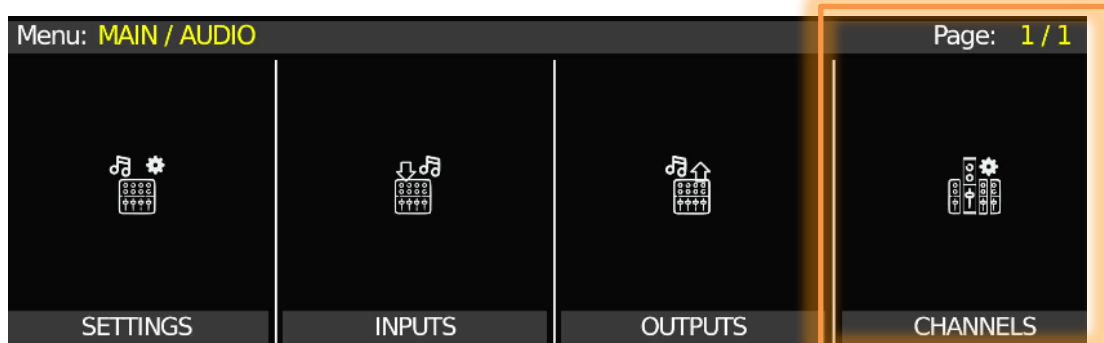
EXT 2 INPUT assigns an input source to the second external monitoring selection. When a monitor output is configured to reproduce **EXT 2**, it hears the source assigned under EXT 2 INPUT. Depending on hardware, INPUT MODE and firmware, the list can include:

- Active **MONO** line-input sources
- Active **STEREO** line-input sources
- Corresponding **ANALOG** input labels
- Additional digital or network sources, when exposed by the installed firmware
- **NONE**, **EMPTY** or an equivalent unassigned state, when available



4. CHANNELS

The **CHANNELS** section configures the audio sources assigned to the console's channel strips. It can be accessed through: **MAIN → AUDIO → CHANNELS**



The section includes:

- **CHANNEL 1-8 or 1-12 or 1-16** depending on HW console configuration.

Each channel can manage two alternative input sources:

- A primary source, identified as **CH-A**
- A secondary source, identified as **CH-B**

This allows two different audio inputs to share the same channel strip. The operator can switch between them without assigning the sources again.

ATTENTION: The GAIN encoder associated with the channel can normally be pressed to switch between CH-A and CH-B. Rotating the encoder adjusts the gain of the currently selected source.

CHANNEL 1-8

The **CHANNEL 1** section configures the sources and processing pages associated with the first channel strip of the console. It can be accessed through: **MAIN → AUDIO → CHANNELS → CHANNEL 1**

CHANNEL 1 can manage two alternative input sources:

- A primary source, identified as **CH-A**
- A secondary source, identified as **CH-B**

Only one of these sources is normally active on CHANNEL 1 at a time. The operator can switch between them without having to change the channel assignment manually. The CHANNEL 1 page includes:

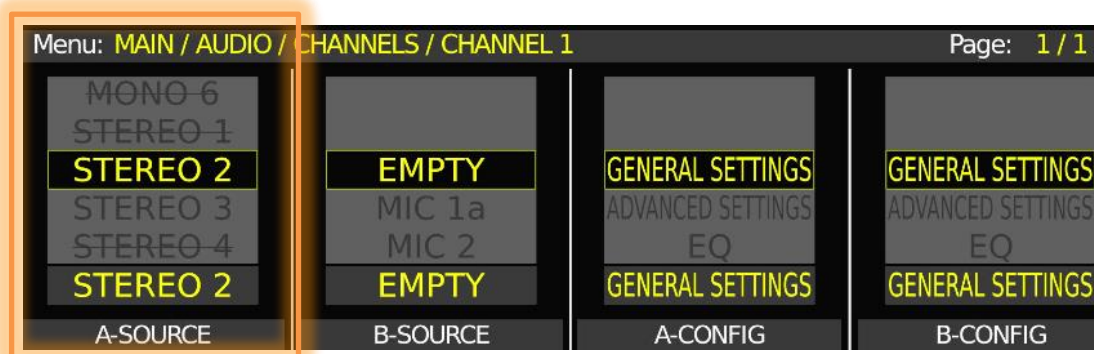
- **A-SOURCE**
- **B-SOURCE**
- **A-CONFIG**
- **B-CONFIG**

A-SOURCE and B-SOURCE assign the two inputs available on CHANNEL 1. A-CONFIG and B-CONFIG provide access to the processing and operating parameters of the corresponding assigned source.

Important: CHANNEL 1 is a channel-strip position, while A-SOURCE and B-SOURCE are the two audio inputs that can be assigned to it.

INPUTS SETTING (CHANNEL 1-8)

A-SOURCE:



A-SOURCE assigns the primary audio input to CHANNEL 1. When CH-A is active: The channel display shows the A-SOURCE name - The GAIN encoder controls the gain associated with A-SOURCE - A-CONFIG provides access to its settings - The CHANNEL 1 fader controls its output level. In A-SOURCE you can select all current INPUT MODE. **Important:** only sources displayed in the A-SOURCE list can be assigned to CHANNEL 1.

B-SOURCE:



B-SOURCE assigns the secondary or alternative input to CHANNEL 1. When CH-B is active: The CHANNEL 1 display shows the B-SOURCE name - The GAIN encoder controls the gain associated with B-SOURCE - B-CONFIG provides access to its parameters - The CHANNEL 1 fader controls B-SOURCE. B-SOURCE can normally use the same available source categories as A-SOURCE. **Important:** B-SOURCE normally replaces A-SOURCE when CH-B is selected. It is not automatically mixed with A-SOURCE.

A-CONFIG:

A-CONFIG opens the configuration of the source assigned to A-SOURCE (It is a shortcut).

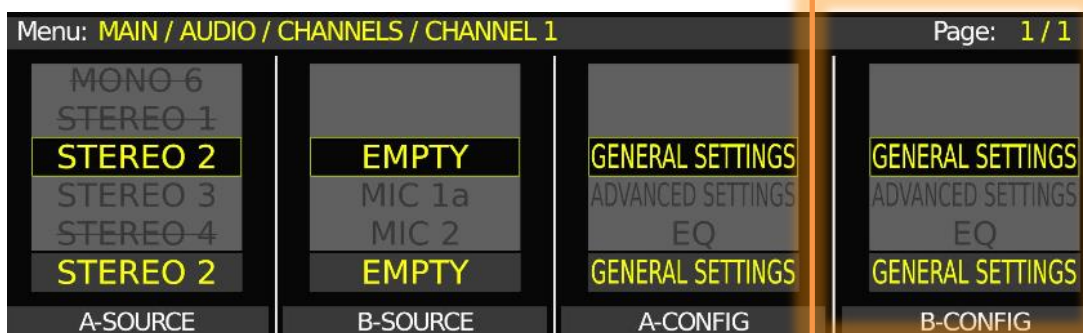
It can be accessed through: **MAIN → AUDIO → CHANNELS → CHANNEL 1 → A-CONFIG**

The selectable configuration sections are:

- **GENERAL SETTINGS**
- **ADVANCED SETTINGS**
- **EQ**
- **DUCKING**
- **DYN**
- **MR LIGHT**

A-CONFIG does not assign the source. The input must first be selected through A-SOURCE. The available configuration sections depend on the assigned source type. Unsupported sections are normally disabled or not selectable. **Important:** A-CONFIG modifies the input source assigned to CH-A. If that source is also assigned to another channel, its source-related processing can be affected there as well.

ATTENTION: For further information about the options available in this section, refer to the corresponding category under **INPUTS**.

B-CONFIG:

B-CONFIG opens the configuration of the source assigned to B-SOURCE. It can be accessed through:

MAIN → AUDIO → CHANNELS → CHANNEL 1 → B-CONFIG

The selectable configuration sections are:

- **GENERAL SETTINGS**
- **ADVANCED SETTINGS**
- **EQ**

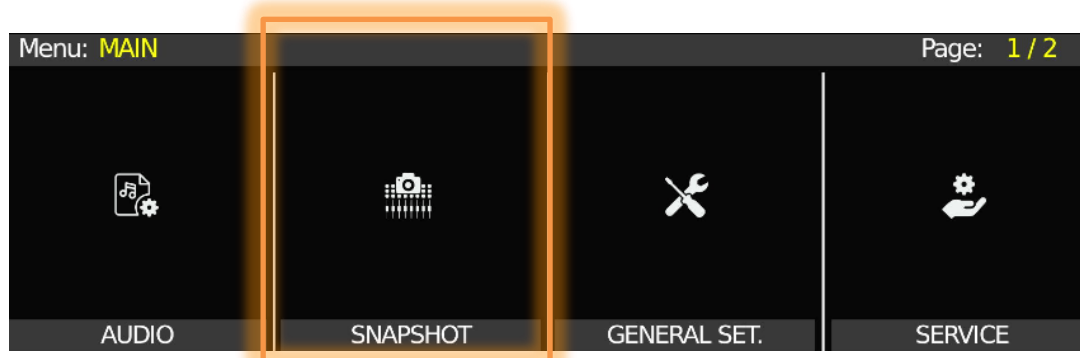
- DUCKING
- DYN
- MR LIGHT

B-CONFIG can be opened while CH-A is active. This allows the operator to prepare the alternative source before selecting it on CHANNEL 1. **Important:** B-CONFIG modifies B-SOURCE even when CH-B is not currently selected on CHANNEL 1.

ATTENTION: For further information about the options available in this section, refer to the corresponding category under **INPUTS**.

SNAPSHOT

The **SNAPSHOT** section, available under the **MAIN** menu, allows the operator to **store and recall predefined console operating conditions**. A snapshot can contain channel assignments, processing settings, fader levels, bus routing and other parameters, depending on the selections made in **SETTINGS**. Snapshots are useful for quickly preparing the console for different programmes, studios, presenters or production configurations. The Oxygen 3000 Plus provides **10 snapshot memory slots** and **four dedicated snapshot recall buttons**. Snapshot data is stored in the console's internal memory and it's not exportable. The **Snapshot** function allows the operator to switch from one console configuration to another while the system is **ON AIR**. Before using this function, each console configuration—together with its assigned sources—must be saved to a separate Snapshot.



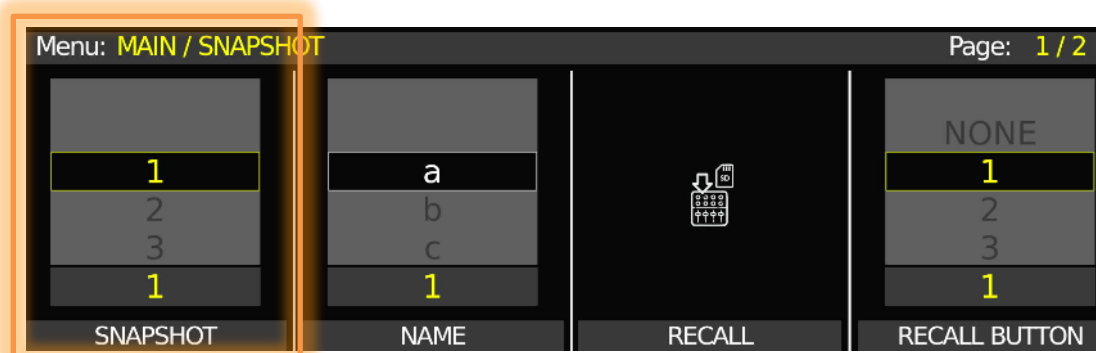
To recall a configuration, press and hold the Snapshot button previously assigned to it. The new configuration will be applied only when all channel faders are fully lowered. If one or more faders remain raised during the transition, the corresponding channels will continue using the previous configuration. Their new settings will become active only after the relevant faders have been moved completely down. This prevents sudden source changes on channels that are currently active **ON AIR**.

SNAPSHOT – Memory selection

Selects the **snapshot memory slot** on which the following operation will be performed. **Selectable options:**

- **1–10:** Selects one of the ten available snapshot memories.

The selected number becomes the target for the **NAME**, **RECALL**, **RECALL BUTTON**, **SAVE**, **DELETE** and **SETTINGS** operations. The **SNAPSHOT** field is shown on both menu pages so that the operator can select the required memory without returning to the previous page. Both fields refer to the **same snapshot memory bank**.



INPUTS SETTING (SNAPSHOT)

NAME:



Assigns a customised name to the selected snapshot. A clear name helps the operator identify the required configuration before recalling it. The label can contain up to **8 characters**.

RECALL:

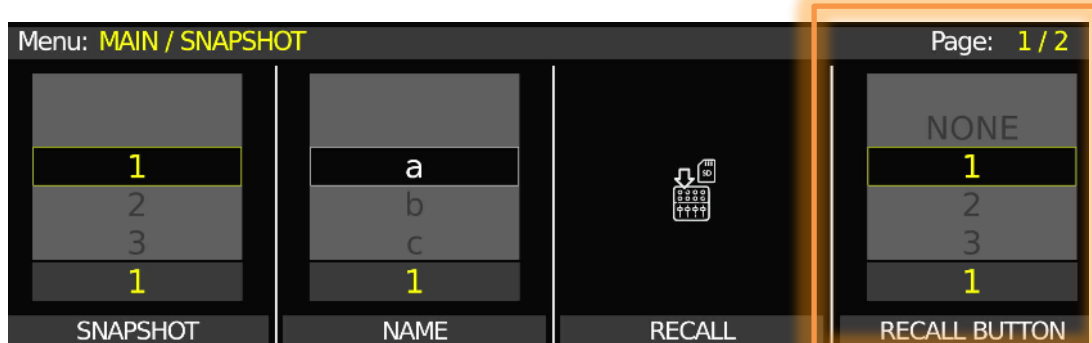


Loads the data stored in the selected snapshot memory and applies it to the console. **Operation:**

- Select the required **SNAPSHOT number**.
- Select **RECALL** to load it.

Only the parameter groups enabled in **SETTINGS** are recalled. Parameters excluded from the snapshot remain unchanged. **Important:** A recall operation can change active channels, fader levels, processing and output routing. Use this function carefully when the console is **ON AIR**.

RECALL BUTTON:



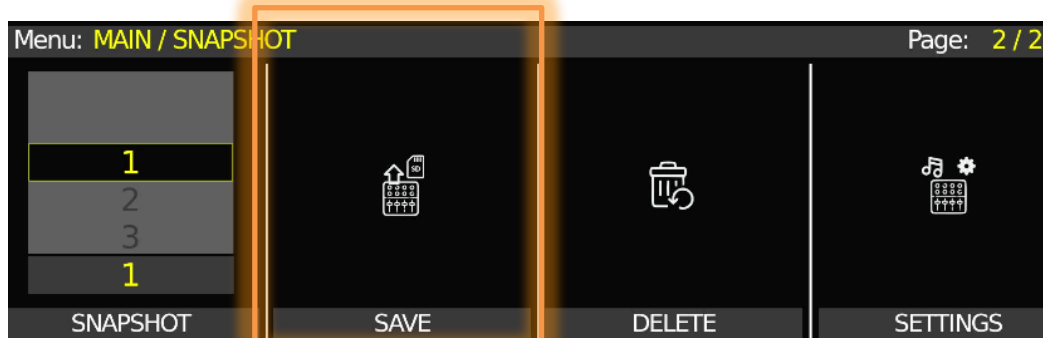
Assigns the selected snapshot to one of the **four dedicated snapshot recall buttons** on the control surface. This allows the operator to recall a frequently used configuration without navigating through the menu.

Selectable options:

- **BUTTON 1-4**

Pressing the assigned physical button recalls the associated snapshot. The four recall buttons are intended to provide **fast access to the most frequently used configurations**.

SAVE:

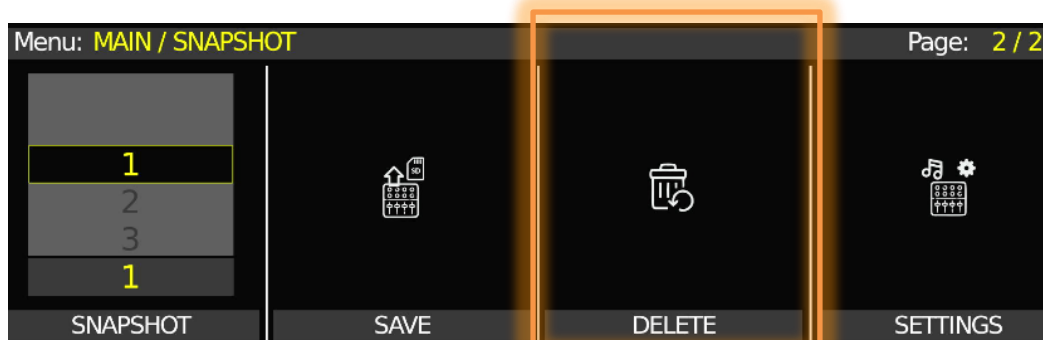


Stores the console's current operating conditions in the selected snapshot memory. **Operation:**

- Select a snapshot slot from **1 to 10**.
- Configure the parameter groups under **SETTINGS**.
- Select **SAVE**.

If the selected slot already contains a snapshot, saving new data will normally **overwrite the previously stored configuration**. Only the parameters enabled under **SETTINGS** are included in the snapshot.

DELETE:



Deletes the data stored in the selected snapshot memory. **Operation:**

- Select the required snapshot slot.
- Select **DELETE**.
- Confirm the operation if requested by the software.

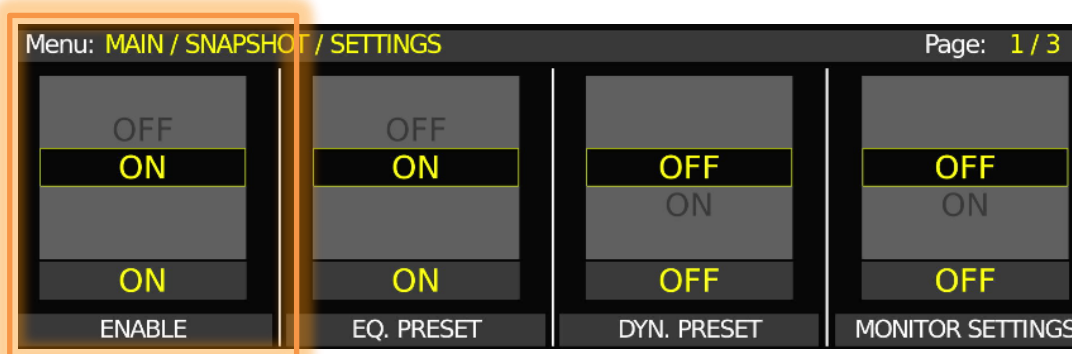
Deleting a snapshot clears the selected memory slot but does not delete the console's general system configuration. If that snapshot was assigned to a **RECALL BUTTON**, the related button will no longer be able to recall the deleted configuration until a valid snapshot is assigned. **Important:** Verify the selected snapshot number before confirming the deletion.

ADVANCED SETTINGS:



The **ADVANCED SETTINGS** section determines which groups of console parameters are applied when a snapshot is recalled. Each parameter can be enabled or excluded independently, allowing the operator to create either a **complete snapshot** or a **partial snapshot**. For example, a snapshot may recall only the **EQ and dynamics presets** without changing fader levels, channel status or output routing.

ENABLE:



Globally enables or disables the application of the snapshot's selected parameter groups.

- **ON:** Snapshot recall is enabled and the console applies the parameter groups selected in **ADVANCED SETTINGS**.
- **OFF:** Application of the snapshot settings is disabled or bypassed.

This parameter acts as the **master control for the snapshot recall configuration**.

EQ PRESET:

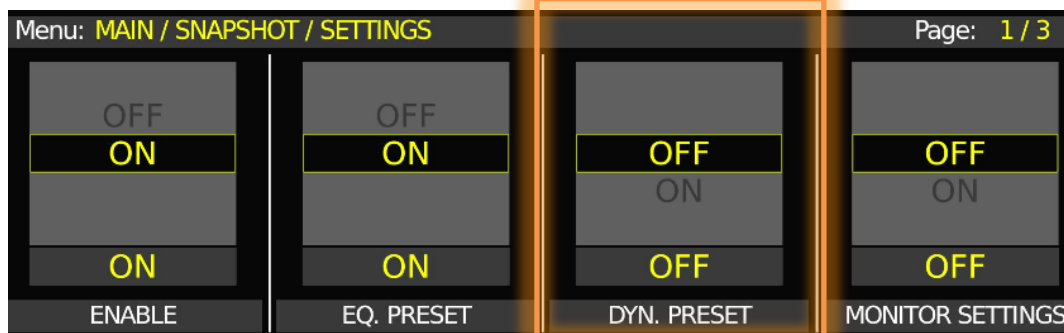


Determines whether the channel **equalisation configurations** contained in the snapshot are recalled.

- **ON:** The stored EQ presets and their related channel configurations are applied.
- **OFF:** Current EQ settings remain unchanged.

This function is useful when different programmes or presenters require specific tonal adjustments.

DYN PRESET:



Determines whether the stored **dynamic-processing presets** are recalled. These may include compressor, limiter, expander or noise-gate configurations, according to the processing available for the selected source.

- **ON:** The stored dynamics presets are applied.
- **OFF:** Current dynamics settings remain unchanged.

This allows the required dynamic processing to be restored without manually configuring each channel.

MONITOR SETTINGS:

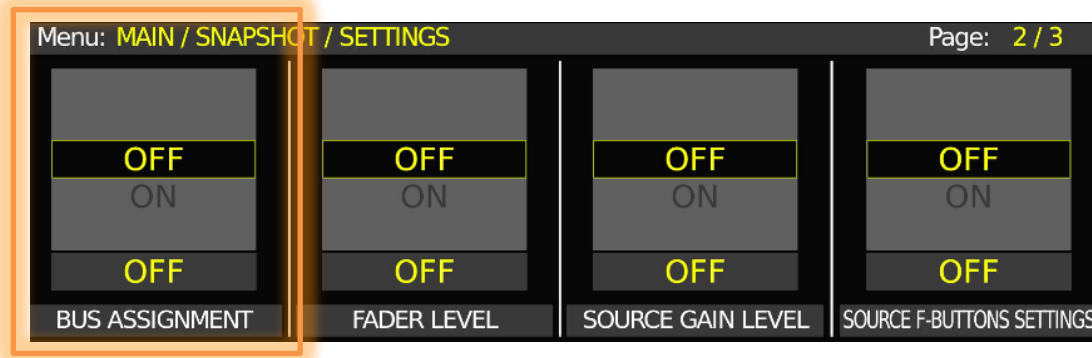


Determines whether the console's **monitoring configuration** is recalled. This may include Control Room and Studio monitoring selections, listening modes and other related monitor parameters supported by the installed firmware.

- **ON:** Stored monitor settings are applied.
- **OFF:** Current monitor settings remain unchanged.

Important: Enabling this option may change what the operator or studio personnel hear through the monitor loudspeakers and headphones.

BUS ASSIGNMENT:

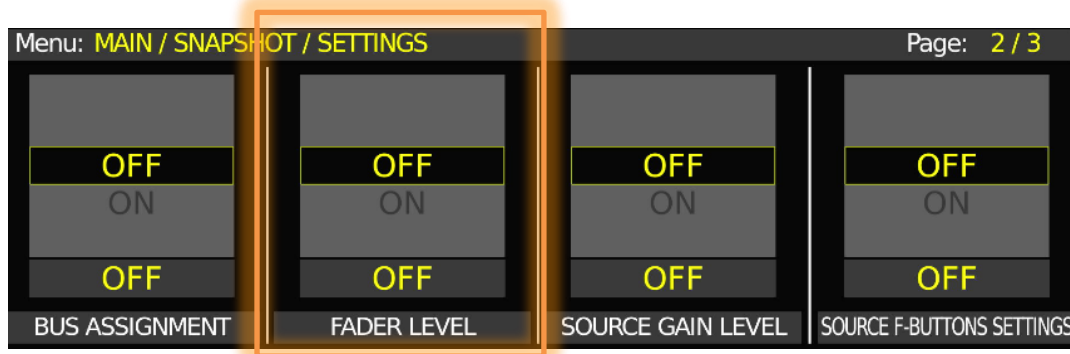


Determines whether the routing of each channel to the console buses is recalled. The affected assignments may include **PGM**, **SUB**, **AUX 1** and **AUX 2**.

- **ON:** Stored bus assignments are restored.
- **OFF:** Current channel-to-bus routing remains unchanged.

Important: Enabling this option may add or remove a channel from an ON-AIR programme bus.

FADER LEVEL:

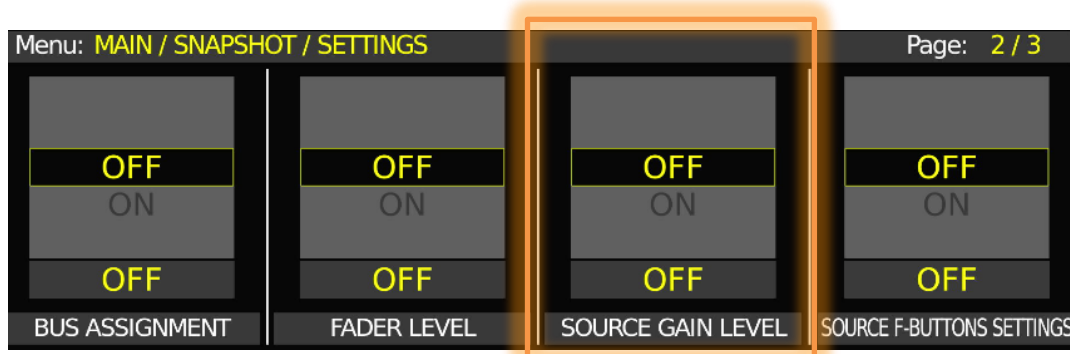


Determines whether the channel attenuation or level values controlled by the faders are recalled.

- **ON:** Stored fader levels are applied.
- **OFF:** Current fader levels remain unchanged.

This option is useful for restoring a predefined mix balance. It should be used carefully during live operation because it may produce an **immediate change in programme levels**.

SOURCE GAIN LEVEL:

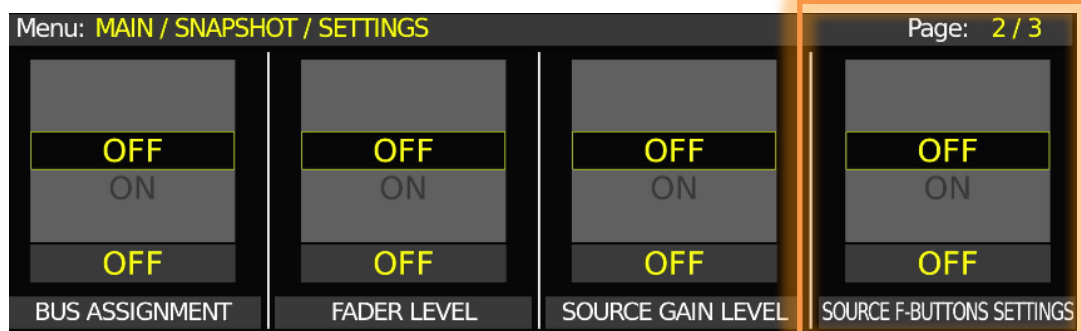


Determines whether the gain settings associated with the assigned input sources are recalled.

- **ON:** Stored source gain levels are applied.
- **OFF:** Current source gain levels remain unchanged.

Unlike **FADER LEVEL**, which controls the channel's mix level, **SOURCE GAIN LEVEL** adjusts the level of the assigned source before the channel fader stage.

SOURCE F-BUTTONS SETTINGS:



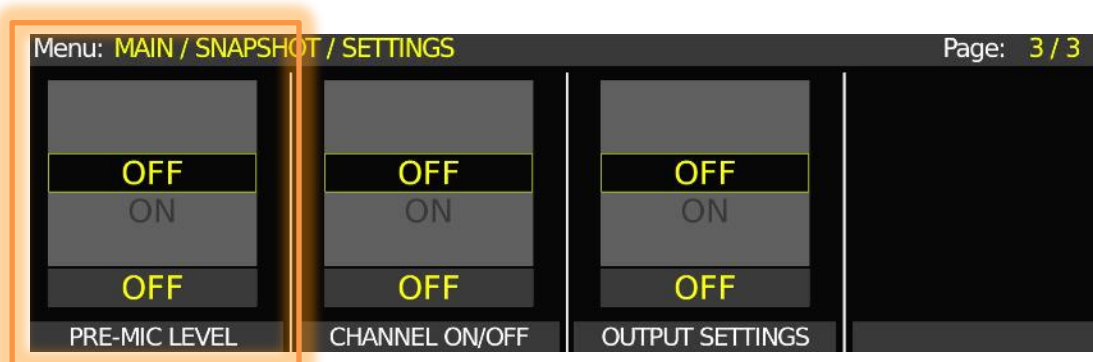
Determines whether the configurations assigned to the channel **F1 and F2 buttons** are recalled.

Depending on the selected input source, these buttons may control functions such as **EQ, compressor, noise gate, ducking, talkback** or other source-specific operations.

- **ON:** Stored F1 and F2 button assignments are restored.
- **OFF:** Current F1 and F2 configurations remain unchanged.

This allows the snapshot to restore the required quick-access functions for each source.

PRE-MIC LEVEL:



Determines whether the microphone **preamplifier gain levels** are recalled. This is the analogue input gain applied before analogue-to-digital conversion.

- **ON:** Stored microphone preamplifier levels are applied.
- **OFF:** Current microphone preamplifier levels remain unchanged.

Important: Changes to preamplifier gain directly affect microphone sensitivity, signal level and available headroom. Excessive gain may cause clipping or feedback.

CHANNEL ON/OFF:

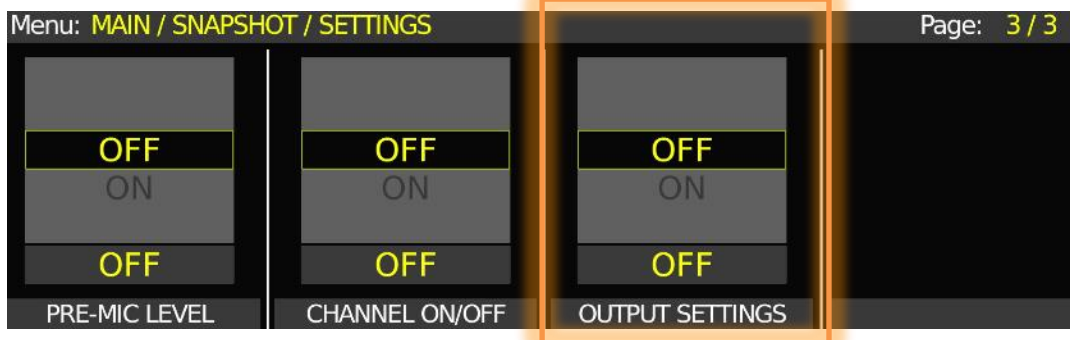


Determines whether the operating status of each physical or virtual channel is recalled.

- **ON:** Channels are restored to their stored **ON** or **OFF** condition.
- **OFF:** Current channel activation states remain unchanged.

Important: When enabled, recalling a snapshot may immediately open or mute channels. The operator should verify the snapshot before recalling it while the console is **ON AIR**.

OUTPUT SETTINGS:



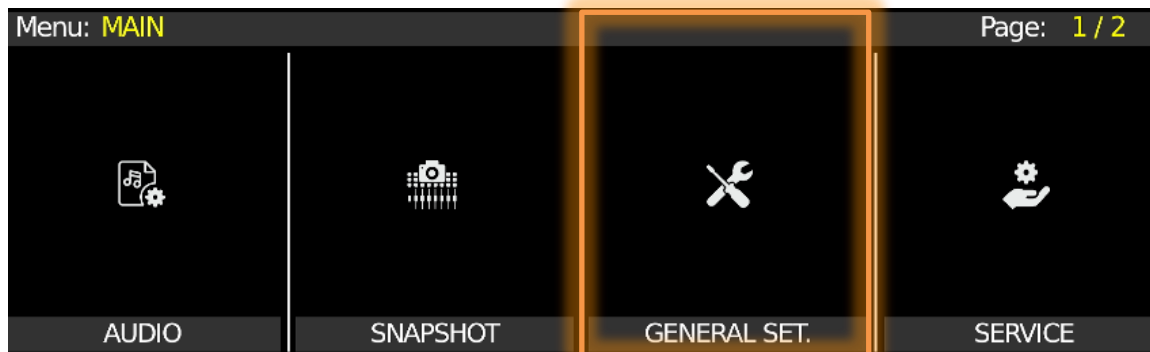
Determines whether the stored configuration of the console outputs is recalled. Depending on the firmware and output type, this may include **output source selection, gain, operating mode, routing and other output-related parameters**.

- **ON:** Stored output settings are applied.
- **OFF:** Current output settings remain unchanged.

This function is useful when different productions require different output destinations or routing configurations. **Important:** Enabling **OUTPUT SETTINGS** may change the signals delivered to transmitters, recording systems, streaming devices or external studio equipment.

GENERAL SETTINGS

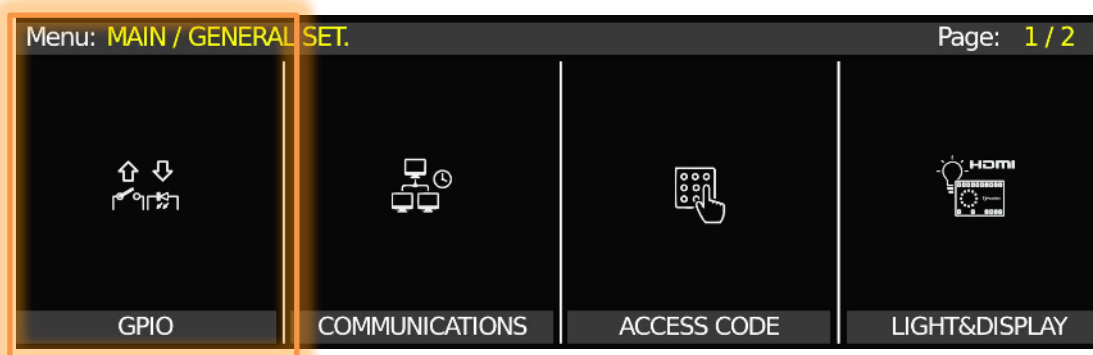
The **GENERAL SETTINGS** section contains the system-level configuration parameters of the Oxygen 3000 Plus. Unlike the AUDIO menu, these settings do not directly process an input signal; they configure the console's **external controls, network communication, security, lighting, displays and automation functions**.



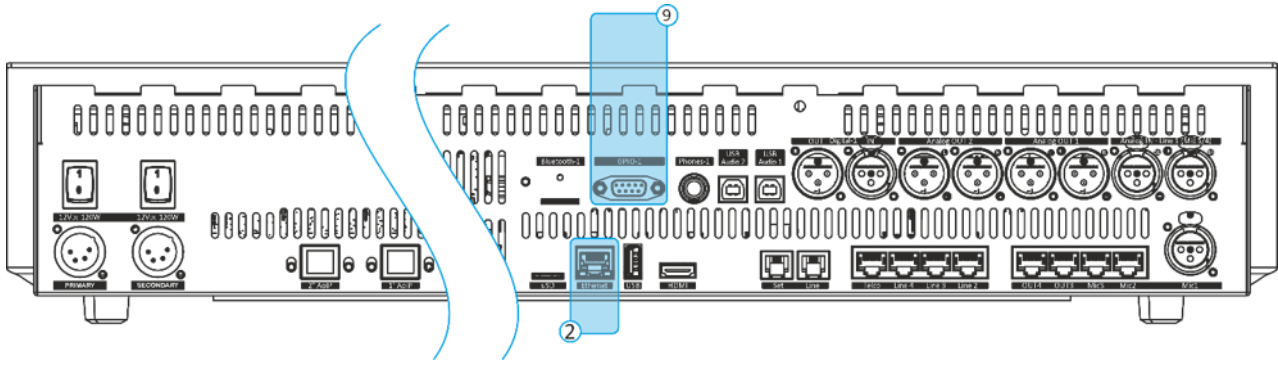
The available sections are:

- **GPIO**
- **COMMUNICATIONS**
- **ACCESS CODE**
- **LIGHT & DISPLAY**
- **MR LIGHT**
- **DMX**
- **SMART KEYS**
- **ON AIR TRIGGERS**

1. GPIO – GENERAL-PURPOSE INPUTS AND OUTPUTS



Here we will talk about only the physical GPIO connections. You can find the GPIO over IP in the TELCO part.] The full version of OXYGEN comes with two SUB-D 9pin (female DE-09) as a connector of general-purpose input/output (GPIO). Each one of these two connectors is equipped with two programmable GPI and two programmable GPO.

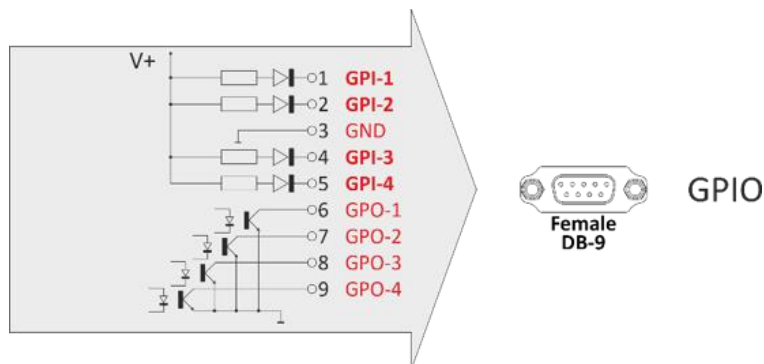


The **GPIO** section is used to configure the physical control signals exchanged between the OXYGEN 3000 Plus and external broadcast equipment.

- A **GPI — General-Purpose Input** receives a contact closure or logic signal from an external device and uses it to execute a console command.
- A **GPO — General-Purpose Output** sends a control or status signal from the console to external equipment.

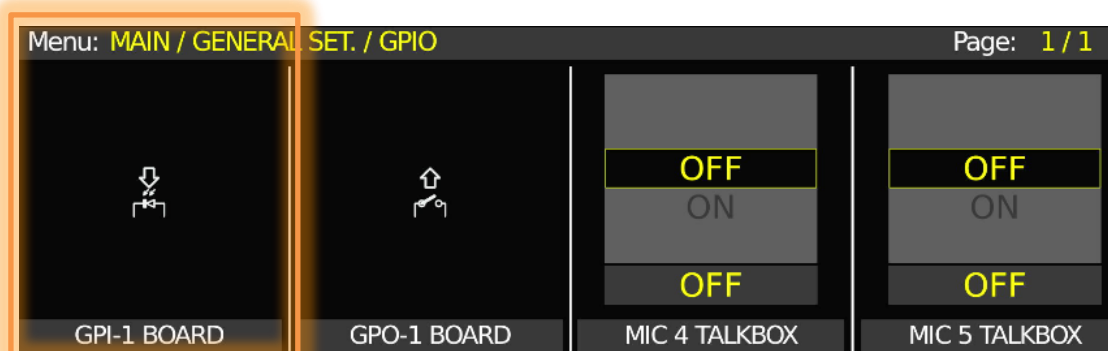
GPIO signals do not transport audio. They are normally used to control **On Air lights**, **Talkback functions**, **telephone hybrids**, **microphone activation**, **automation systems** and other studio devices.

MAIN / GENERAL SETTINGS / GPIO



GPI-1 BOARD:

The **GPI-1 BOARD** menu configures the physical General-Purpose Inputs associated with the **first Audio I/O Board** of the OXYGEN 3000 Plus. A **GPI — General-Purpose Input** receives an electrical control signal or contact closure from an external device. When the input is activated, the console executes the function assigned to that GPI. These connections carry **control signals only** and **do not transport audio**.



The GPI-1 BOARD section includes:

- **GPI-1**
- **GPI-2**
- **GPI-3**
- **GPI-4**
- **GPI TELCO 1**

The first four inputs are available on the board's **female DE-09 connector**, while GPI TELCO 1 is carried by the dedicated **TELCO I/O RJ45 connection**.

ATTENTION: It should be noted that the availability of the **GPI** and **GPO** parameters depends on the **MIC TALKBOX** configuration. When **MIC 4–5 TALKBOX** is set to **ON**, all options in the **GPI-1 BOARD** section are locked, except **GPI TELCO 1**. In the **GPO-1 BOARD** section, all options remain editable except **GPO 1** and **GPO 2**, which are locked. When **MIC 4–5 TALKBOX** is set to **OFF**, all options in both the **GPI-1 BOARD** and **GPO-1 BOARD** sections can be modified. Similarly, when **MIC 9–10 TALKBOX** is set to **ON**, all options in the **GPI-2 BOARD** section are locked, except **GPI TELCO 1**. In the **GPO-2 BOARD** section, all options remain editable except **GPO 1** and **GPO 2**, which are locked. When **MIC 9–10 TALKBOX** is set to **OFF**, all options in both the **GPI-2 BOARD** and **GPO-2 BOARD** sections can be modified.



INPUTS SETTING (GPI-1 BOARD)

Selectable GPI assignments

Each general-purpose input can be independently associated with a console command. The available list is generated according to the **installed firmware and current console configuration**.

All GPI inputs can be assigned to a wide range of functions, including:

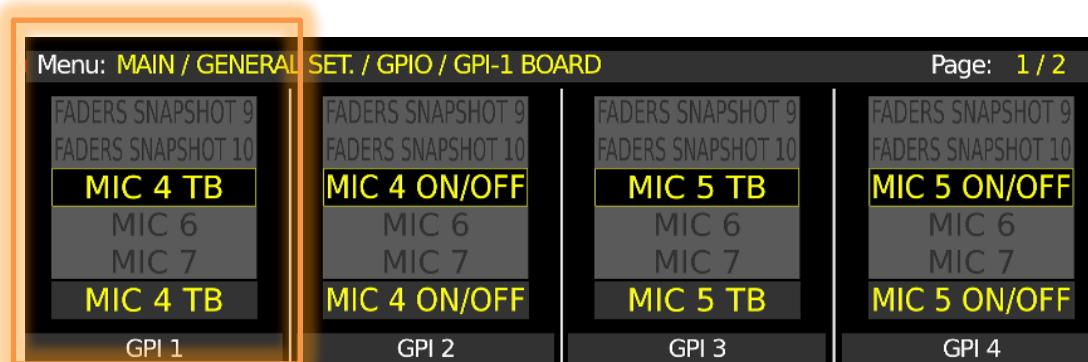
- **Audio input sources**, such as MIC, LINE, DIGITAL or other configured channels: used to control the ON/OFF status of the associated source.

- **Snapshots:** allows recalling a stored console configuration when the GPI is activated.
- **TELCO RING commands**, such as RING 1, RING 2 or other available TELCO channels: used to signal incoming calls from external telephone hybrids.
- **Dante sources or events**, when available in the system configuration.
- **Talkback functions:** activates the corresponding Talkback path.
- **NONE or UNASSIGNED:** disables the input assignment, when this option is available.

The exact list of selectable functions depends on the **firmware version and system configuration**, including enabled sources, TELCO channels and network audio options.

Only **one command can normally be assigned to each physical GPI**.

GPI-1:



GPI-1 configures the first programmable control input on the first board's DE-09 connector.

It can be connected to an external button, relay contact, telephone hybrid or automation system. When the external signal becomes active, the console executes the function selected in the GPI-1 assignment field. On the documented DE-09 pinout, **GPI-1 is available on pin 1**.

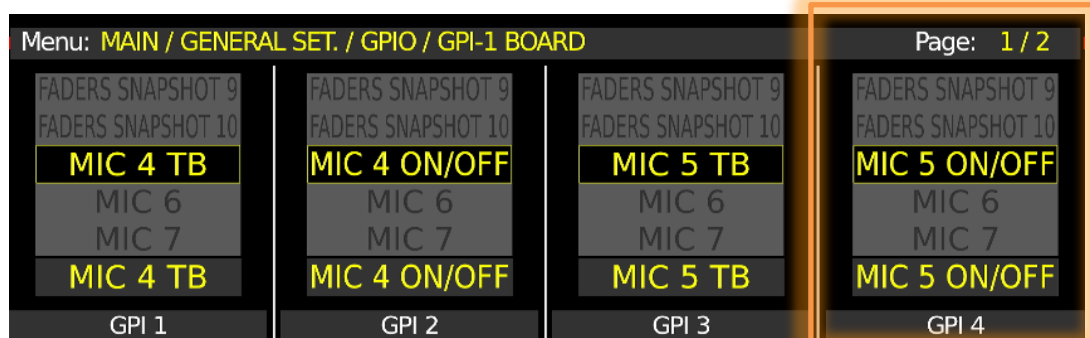
GPI-2:



GPI-2 configures the second programmable control input. It operates independently from GPI-1 and can therefore be assigned to a different function, such as a snapshot recall, a Dante-related command, a telephone Ring signal or channel ON/OFF control. On the documented DE-09 pinout, **GPI-2 is available on pin 2**.

GPI-3:

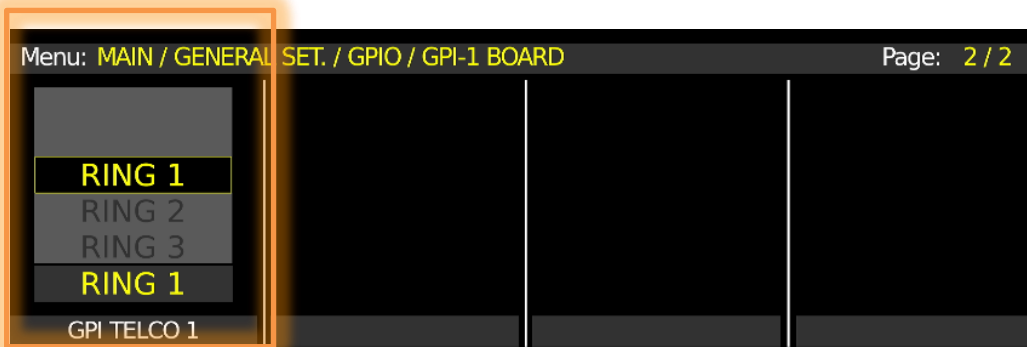
GPI-3 configures the third programmable input of the board. This input can receive a control command from an external studio device and associate it with any compatible function available in the assignment menu, including sources, snapshots, TELCO events or Dante-related functions. On the documented DE-09 pinout, **GPI-3 is available on pin .**

GPI-4:

GPI-4 configures the fourth programmable input of the board.

Like the other general-purpose inputs, it can be assigned to any available function supported by the current firmware, including audio sources, snapshots, TELCO Ring signals, Talkback or Dante-related events. On the documented DE-09 pinout, **GPI-4 is available on pin 5.** **Note:** pin 3 of the DE-09 connector is the reference **GND** connection. The external wiring must follow the official Axel Technology GPIO pinout.

GPI TELCO 1:



GPI TELCO 1 configures the dedicated input signal carried by the **TELCO 1 I/O RJ45 connector**. It is separate from the four programmable GPIs available on the DE-09 connector.

Its standard application is to receive the **RING signal from an external telephone hybrid**. When the external device detects an incoming call, it activates this GPI and the console reports the Ring condition on the TELCO 1 channel, normally by making the associated F-button LED flash.

In addition to the standard TELCO RING function, **GPI TELCO 1 can be assigned to the same categories of functions available for the other GPIs**, including:

- Audio input sources
- Snapshots
- TELCO commands
- Dante-related functions
- Talkback functions

For standard external-hybrid operation, the corresponding assignment is:

- **TELCO 1 RING**: associates the physical TELCO 1 GPI with the incoming-call indication for TELCO 1.

Important: GPI TELCO 1 receives the **RING indication** from the external hybrid. The outgoing **HOOK/DROP command** is handled separately by **GPO TELCO 1**.

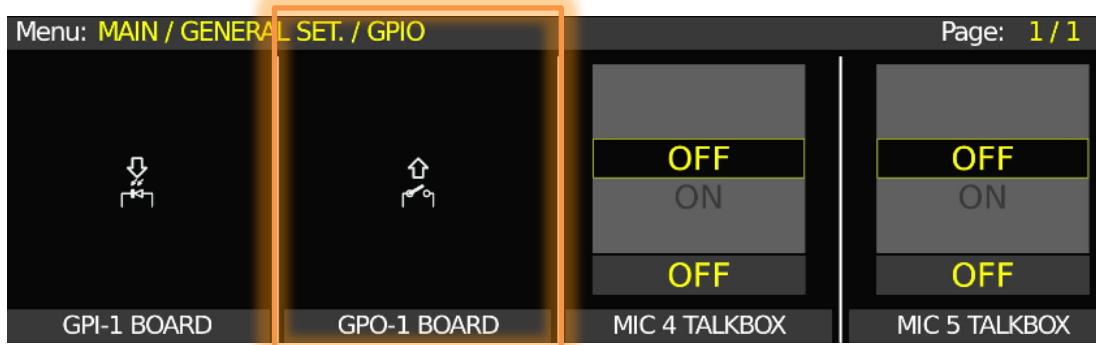
GPI-2 BOARD:

The **GPI-2 BOARD** menu configures the physical General-Purpose Inputs associated with the **second Audio I/O Board** of the OXYGEN 3000 Plus. A **GPI — General-Purpose Input** receives a compatible electrical control signal from an external device. When the signal becomes active, the console executes the command assigned to that particular input. GPI-2 BOARD is available only when the OXYGEN 3000 Plus is equipped with a **second Audio I/O Board**. On single-board configurations, these physical inputs are not installed and the corresponding settings cannot be used. Each input belonging to GPI-2 BOARD can be configured **independently**. Activating one input does not automatically activate the others, and the GPI signals of the second board do not depend on those of GPI-1 BOARD. However, inputs from the two boards can be assigned to related commands when required by the studio workflow. For example, one board may manage the Control Room equipment while the other controls a second studio, additional telephone hybrids or separate automation commands.

ATTENTION: It should be noted that all options available within the **GPI-1 BOARD** section are also available within the **GPI-2 BOARD** section.

GPO-1 BOARD:

The **GPO-1 BOARD** menu configures the physical General-Purpose Outputs associated with the **first Audio I/O Board** of the OXYGEN 3000 Plus. A **GPO — General-Purpose Output** sends an electrical control or status signal from the console to external equipment. The output changes its state when the assigned console event becomes active.



INPUTS SETTING (GPO-1 BOARD)

Selectable GPO assignments

Each GPO can normally be associated with one control function.

- **NONE or UNASSIGNED:** disables the output, when this option is available.
- **HOOK 1, HOOK 2 or another available TELCO HOOK command:** controls the Hook/Drop status of an external telephone hybrid.
- **CR-ONAIR:** activates an external Control Room On Air indicator.
- **ST-ONAIR:** activates an external Studio On Air indicator.
- **CONTROL ROOM LIGHT:** controls a warning light associated with the Control Room.
- **ALWAYS ON:** forces the output to remain permanently active.
- **ALWAYS OFF:** forces the output to remain permanently inactive.
- **DANTE TELCO ON / DANTE TELCO OFF:** controls the status of Dante-based telephone interfaces, enabling or disabling the corresponding Telco function over the Dante network.

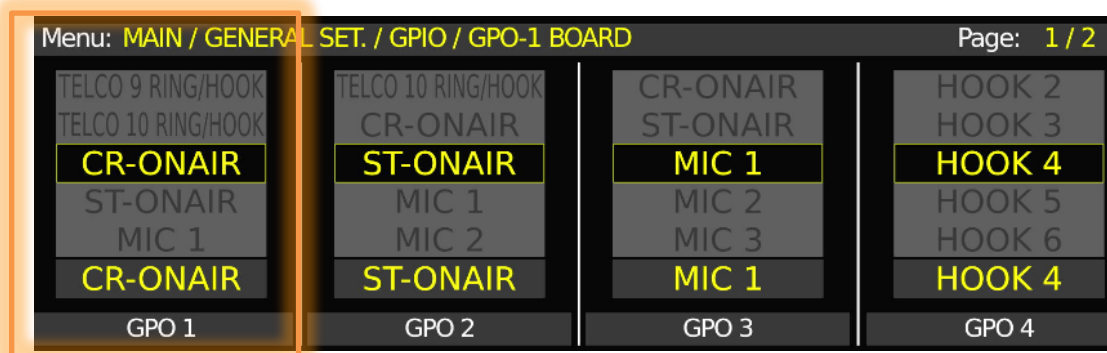
In addition to these functions, the assignment list may also include **audio input sources** available in the console configuration. These can include:

- **MIC inputs**
- **LINE inputs**
- **Dante sources**
- **Streamer inputs**
- **USB audio sources**
- Other enabled input channels

When one of these sources is selected, the GPO is typically activated according to the operational status of that source (for example, when the channel is opened, goes On Air, or meets the configured activation condition). The complete list of assignments may vary according to the **firmware version, enabled sources and console hardware configuration**.

ATTENTION: It should be noted that the availability of the **GPI** and **GPO** parameters depends on the **MIC TALKBOX** configuration. When **MIC 4–5 TALKBOX** is set to **ON**, all options in the **GPI-1 BOARD** section are locked, except **GPI TELCO 1**. In the **GPO-1 BOARD** section, all options remain editable except **GPO 1** and **GPO 2**, which are locked. When **MIC 4–5 TALKBOX** is set to **OFF**, all options in both the **GPI-1 BOARD** and **GPO-1 BOARD** sections can be modified. Similarly, when **MIC 9–10 TALKBOX** is set to **ON**, all options in the **GPI-2 BOARD** section are locked, except **GPI TELCO 1**. In the **GPO-2 BOARD** section, all options remain editable except **GPO 1** and **GPO 2**, which are locked. When **MIC 9–10 TALKBOX** is set to **OFF**, all options in both the **GPI-2 BOARD** and **GPO-2 BOARD** sections can be modified.

GPO-1:



GPO-1 configures the first programmable output associated with the first Audio I/O Board. It can provide an operating-status signal to an external relay, telephone hybrid, warning lamp controller or automation interface. On the documented female DE-09 GPIO pinout, **GPO-1 is available on pin 6**.

GPO-2:



GPO-2 configures the second programmable output. It operates independently from GPO-1 and may be assigned to a different function, such as another telephone Hook command or a separate On Air indication. On the documented DE-09 pinout, **GPO-2 is available on pin 7**.

GPO-3:

GPO-3 configures the third programmable output of the board.

It can be used for an additional studio status, On Air light, telephone hybrid or compatible external control device. On the documented DE-09 pinout, **GPO-3 is available on pin 8.**

GPO-4:

GPO-4 configures the fourth programmable output.

Like the other general-purpose outputs, it sends the status of the assigned console function to an external system. On the documented DE-09 pinout, **GPO-4 is available on pin 9.**

OUT 3:

OUT 3 configures the dedicated GPO signal available through the **ANALOG OUT 3 RJ45 connector.**

This control signal is separate from the analogue audio carried by the same connector. It can be used to control an external indicator, relay interface or other studio equipment according to the selected assignment. **Important:** although the OUT 3 connector carries both audio and GPIO signals, the GPO setting affects only the control line and does not modify the audio routed to ANALOG OUT 3.

OUT 4:



OUT 4 configures the dedicated GPO available through the **ANALOG OUT 4 RJ45 connector**.

It provides the same type of control functionality as OUT 3 but can be assigned and operated independently. The audio output and its associated GPO remain two separate signals carried by the same RJ45 connection.

GPO TELCO 1:



GPO TELCO 1 configures the dedicated control output carried by the **TELCO 1 I/O RJ45 connector**.

Its standard function is to send the **HOOK/DROP command** to an external telephone hybrid:

- **HOOK:** answers or engages the telephone line.
- **DROP:** releases the telephone line.

When the channel F-button is configured in **GPIO mode**, pressing it can activate GPO TELCO 1 to answer the call. Pressing the button again releases the connection.

The corresponding incoming **RING indication** is received separately through **GPI TELCO 1**.

Interaction with MIC 4–5 TALKBOX

When **MIC 4–5 TALKBOX** is set to **ON**, the console reserves the control resources required by the two Talkbox connections. Consequently:

- GPO-1 and GPO-2 are locked and cannot be reassigned.
- GPO-3 and GPO-4 remain configurable.
- OUT 3 and OUT 4 remain configurable.
- GPO TELCO 1 remains configurable.

GPO-2 BOARD:

The **GPO-2 BOARD** menu configures the physical General-Purpose Outputs associated with the **second Audio I/O Board** of the OXYGEN 3000 Plus. A **GPO — General-Purpose Output** sends an electrical control or status signal from the console to compatible external equipment. The output changes its state when the assigned console event becomes active. The GPO signals themselves carry **control information only and do not transport audio**. However, some RJ45 connectors associated with the second Audio I/O Board can carry both audio and dedicated GPO signals. In these cases, the audio and control signals remain electrically and functionally separate. GPO-2 BOARD is available only when the console is equipped with a **second Audio I/O Board**. On single-board configurations, the related physical connections and menu parameters are not available. Each output in GPO-2 BOARD can be configured **independently**. Activating one GPO does not automatically activate the other outputs. GPO-2 BOARD also operates independently from **GPO-1 BOARD**. The two sections control different physical outputs and can therefore be used to manage separate equipment, studios or telephone hybrids.

Menu: MAIN / GENERAL SET. / GPIO / GPO-2 BOARD				Page: 1 / 2
MIC 7 MIC 8 MIC 9 onair TELEPHONE 1 TELEPHONE 2 MIC 9 onair	MIC 7 MIC 8 MIC 10 onair TELEPHONE 1 TELEPHONE 2 MIC 10 onair	HOOK 4 HOOK 5 CR-ONAIR ST-ONAIR MIC 1 CR-ONAIR	HOOK 5 CR-ONAIR ST-ONAIR MIC 1 LINE 1 ST-ONAIR	
GPO 1	GPO 2	GPO 3	GPO 4	

ATTENTION: It should be noted that all options available within the **GPO-1 BOARD** section are also available within the **GPO-2 BOARD** section

Interaction with MIC 9–10 TALKBOX

When **MIC 9–10 TALKBOX** is set to **ON**, the console reserves some of the second board's GPIO resources for the two Talkbox connections. Consequently, **GPO-1 and GPO-2 in GPO-2 BOARD are locked and cannot be reassigned**. The other GPO entries belonging to the second board remain configurable.

When **MIC 9–10 TALKBOX** is set to **OFF**, all the GPO entries in GPO-2 BOARD become available for configuration, including **GPO-1 and GPO-2**.

MIC TALKBOX:

The **MIC TALKBOX** options configure the GPIO resources required by the Talkboxes connected to the dedicated microphone ports of the OXYGEN 3000 Plus. The GPO is activated when the channel associated

with the Talkbox microphone is **ON**. This return signal is used to provide a visual status indication on the Talkbox and to manage the related microphone or On Air indication.

The GPO signal **does not carry audio and does not activate the microphone directly**. Its purpose is to report the console's microphone-channel status to the connected Talkbox.

ON:



When **MIC TALKBOX** is set to **ON**, the console activates the dedicated GPIO configuration for the selected Talkbox connection.

The required GPI and GPO resources are automatically reserved for:

- Receiving the **ON/COUGH button** command.
- Receiving the **TALKBACK button** command.
- Sending the **microphone ON status** to the Talkbox.

Because these GPIO resources are reserved, the corresponding entries in the general-purpose GPI and GPO assignment menus become **locked and cannot be reassigned**.

For the first Audio I/O Board:

- MIC 4 and MIC 5 use the resources associated with **MIC 4–5 TALKBOX**.
- The related **GPO-1 and GPO-2** entries are reserved when the Talkbox function is enabled.

For the second Audio I/O Board:

- MIC 9 and MIC 10 use the resources associated with **MIC 9–10 TALKBOX**.
- The related **GPO-1 and GPO-2 of GPO-2 BOARD** are reserved when the Talkbox function is enabled.

OFF:



When **MIC TALKBOX** is set to **OFF**, the dedicated Talkbox GPIO configuration is disabled.

The corresponding GPI and GPO resources are released and become available for other compatible assignments in the general-purpose GPIO menus. Selecting OFF therefore allows the installer to use these connections for other external control applications when no Talkbox is connected.

Important: MIC TALKBOX ON/OFF primarily controls the allocation of the GPIO resources required by the Talkbox. It must not be confused with the microphone channel's audio ON/OFF status or with the ON MODULE configuration.

2. COMMUNICATIONS

The **COMMUNICATIONS** section contains the network and time-management parameters of the OXYGEN 3000 Plus. These settings allow the console to communicate with external systems and to maintain the correct date and time for functions.



The **COMMUNICATIONS** section includes:

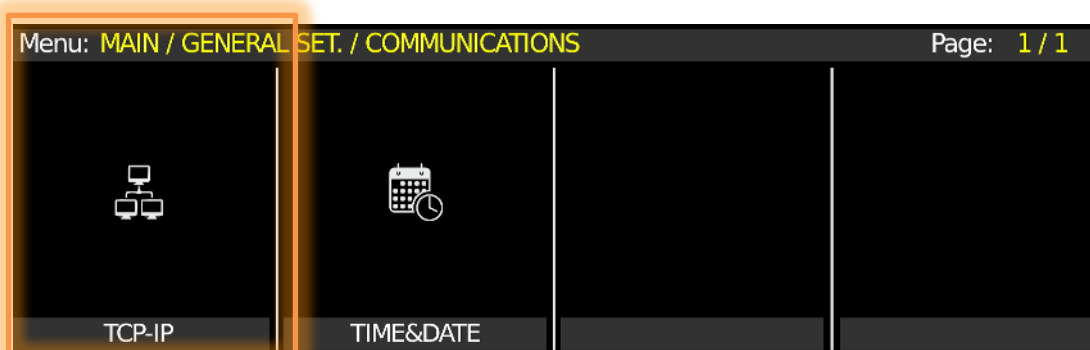
- **TCP-IP**
- **TIME&DATE**

TCP-IP:

The **TCP-IP** section configures the parameters used to connect the console to an Ethernet network. The OXYGEN 3000 Plus supports two configurable network addresses, identified as:

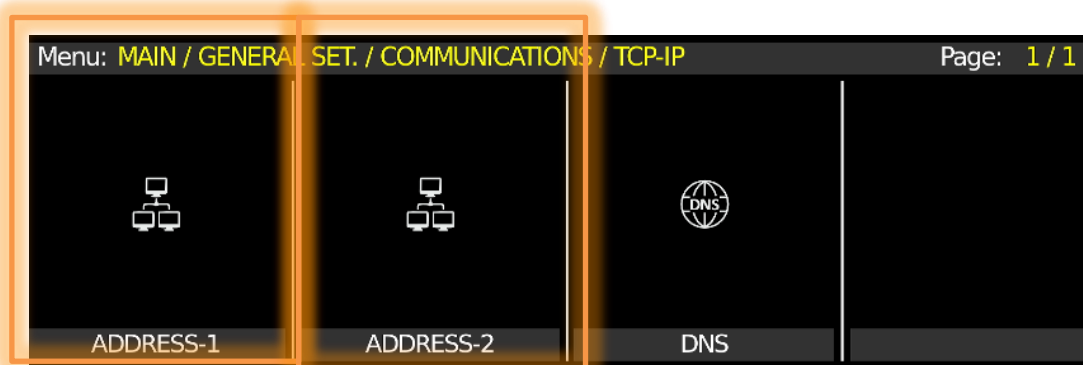
- **ADDRESS-1**
- **ADDRESS-2**

These are two logical IP configurations. They must not necessarily be interpreted as two independent physical Ethernet connectors. One address can be used for standard console management, including the Web interface and OXYGEN Remoter, while the other can be used for communication with compatible external systems.



ADDRESS-1 and ADDRESS-2

ADDRESS-1 and ADDRESS-2 contain the same network parameters and can be configured independently.

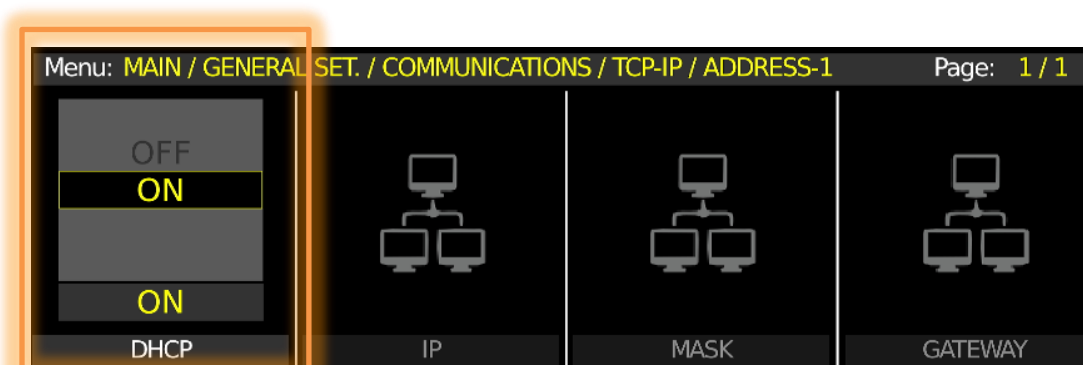


The available parameters are:

- **DHCP**
- **IP**
- **MASK**
- **GATEWAY**

Depending on the installed firmware, the TCP-IP section may also display **DNS** and the console's **MAC** address.

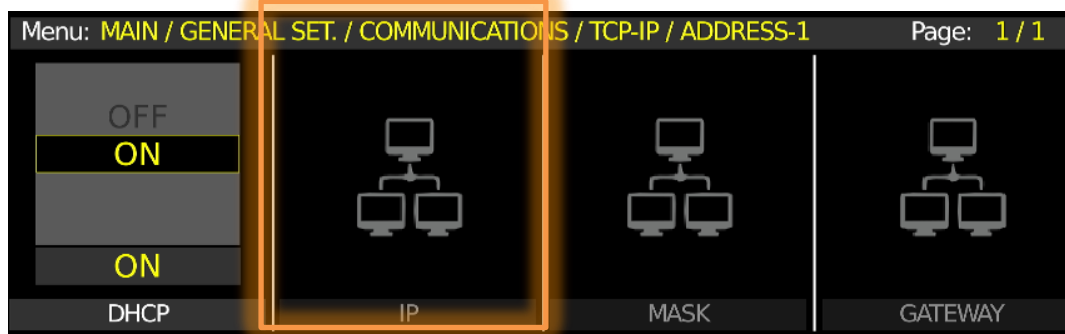
DHCP:



DHCP — Dynamic Host Configuration Protocol determines whether the console obtains its network configuration automatically or uses manually entered values. Selectable options are:

- **ON:** the console requests an IP address and the related network parameters from the network's DHCP server.
- **OFF:** DHCP is disabled and the IP parameters must be entered manually.

DHCP can simplify initial connection, but a **static IP configuration** is normally preferable when the console must always be reached at the same network address.

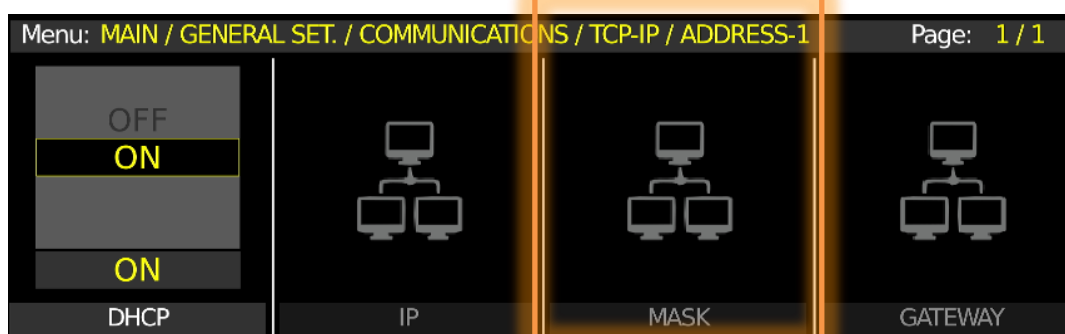
IP:

IP defines the Internet Protocol address assigned to the selected console network configuration. An IPv4 address consists of four numerical fields. Each field can be adjusted from **0** to **255**.

Example: **192.168.120.120**

When DHCP is OFF, the selected IP address must:

- Belong to the correct network.
- Be unique and not already assigned to another device.
- Be compatible with the selected subnet mask.

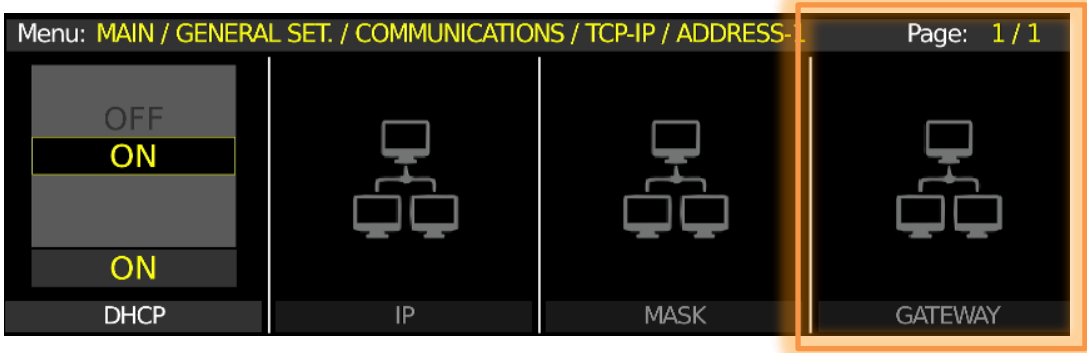
MASK:

MASK defines the subnet mask associated with the selected IP address. It identifies which part of the address represents the local network and which part represents the individual device.

A commonly used value is: **255.255.255.0**

The correct value depends on the network configuration and must be provided by the network administrator.

GATEWAY:

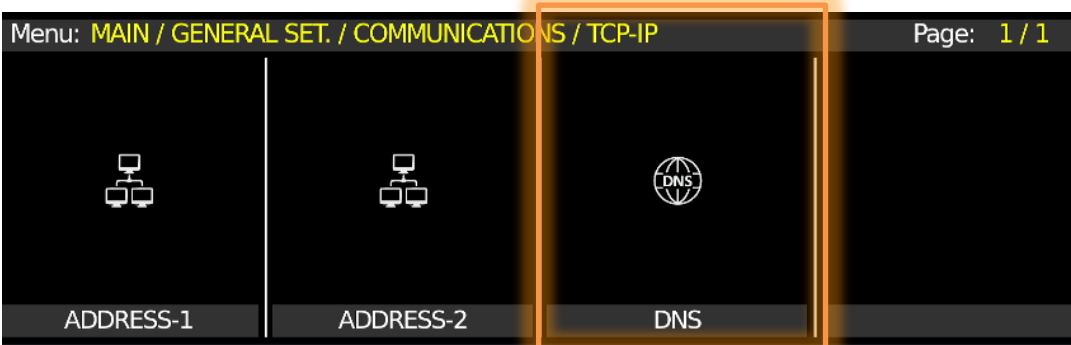


GATEWAY specifies the address of the network device that allows the console to communicate outside its local subnet. The gateway is normally the IP address of the network router. It may be required for:

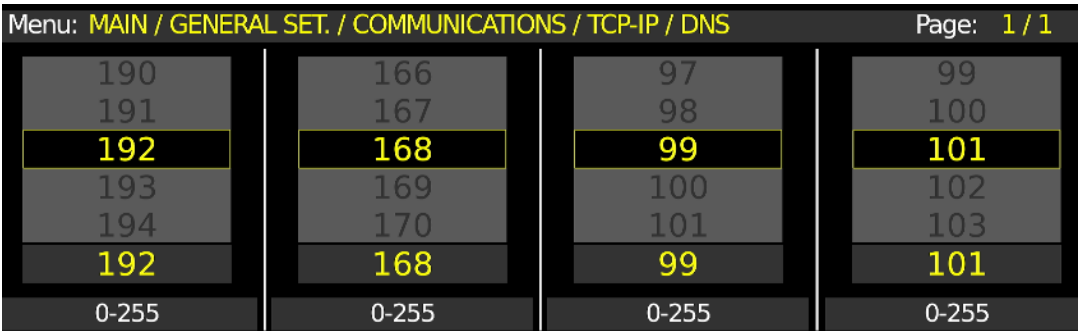
- Internet access
- Online firmware updates
- Remote services outside the local network
- External NTP servers

If the console is used exclusively within the local network, a gateway may not be required.

DNS

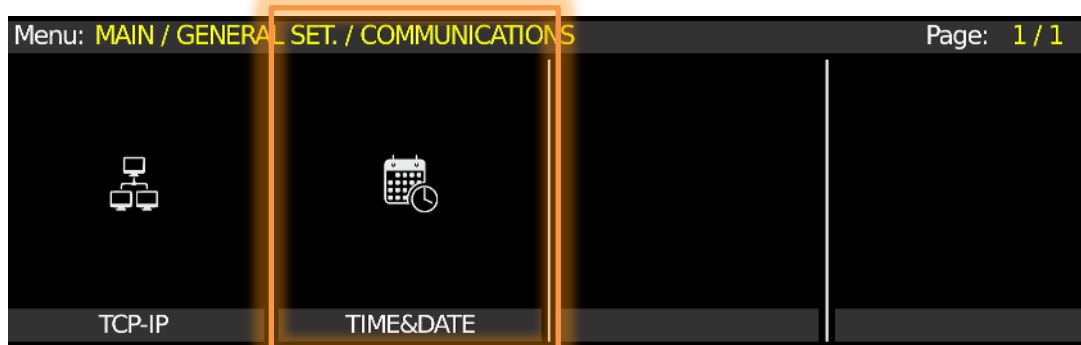


When provided by the installed firmware, **DNS — Domain Name System** specifies the server used to translate host names into IP addresses. DNS is required when the console must reach a network service using a host name instead of a numerical IP address. DNS must be set manually.



TIME&DATE:

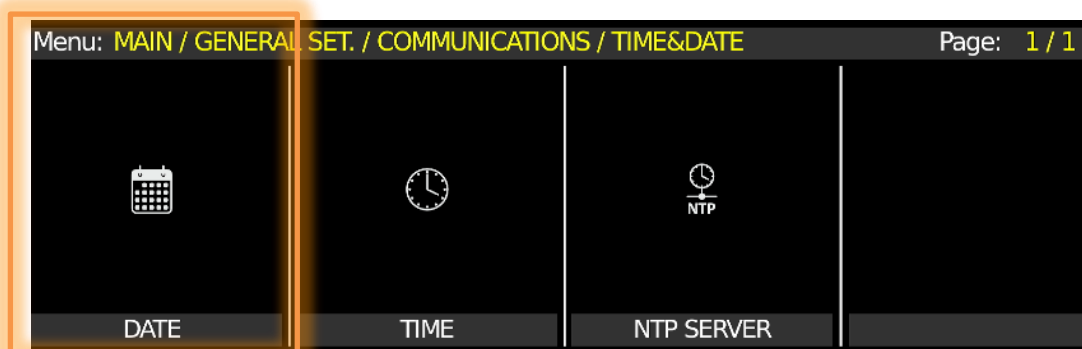
The **TIME&DATE** section configures the internal clock and calendar of the OXYGEN 3000 Plus.



The TIME&DATE section includes:

- **DATE**
- **TIME**
- **TIME ZONE**
- **NTP SERVER**

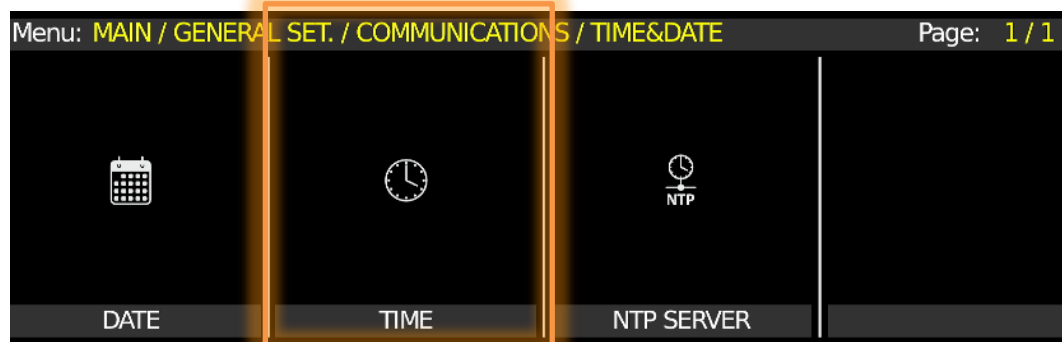
DATE:



DATE sets the console's internal calendar. The editable parameters are:

- **DAY**
- **MONTH**
- **YEAR**

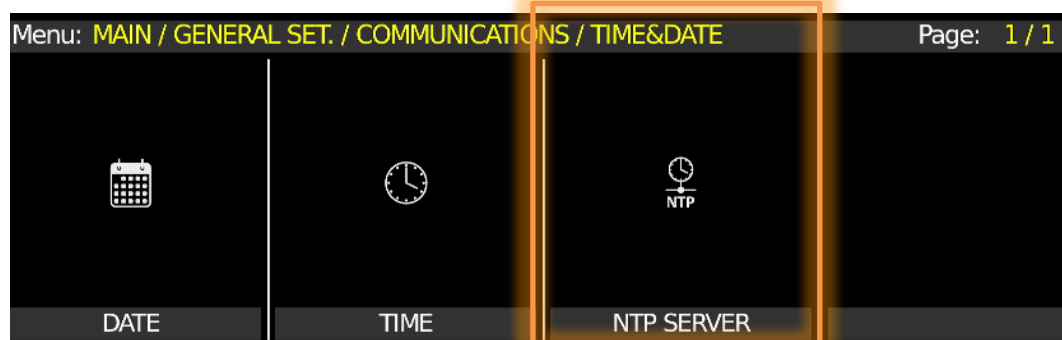
The selected values are used for system logs and other date-dependent console functions.

TIME:

TIME sets the console's internal clock. The editable parameters are:

- **HOURL**
- **MINUTE**
- **SECOND**, when provided by the installed firmware.

When an NTP server is enabled and reachable, the manually configured time may be automatically replaced by the synchronized network time. From the following section, you can set all the parameters related to the date settings. These data parameters are very important for device **LOGS**.

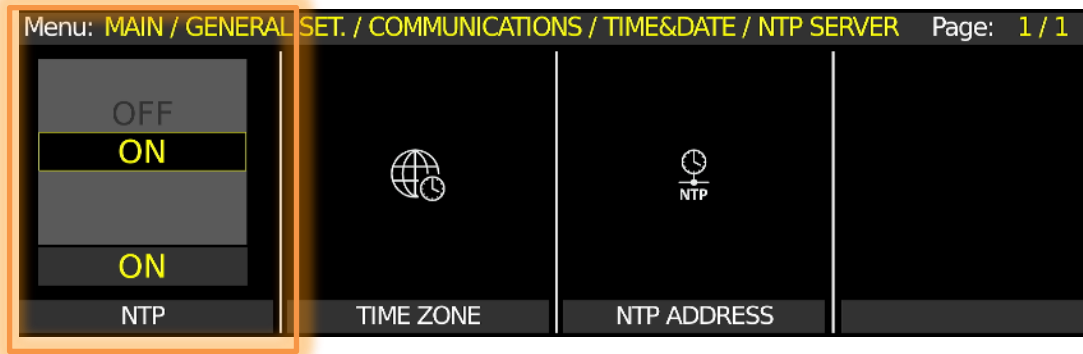
NTP SERVER:

NTP — Network Time Protocol allows the console to synchronize its internal date and time with a network time server. The related parameters normally include:

- **NTP ON/OFF**
- **NTP SERVER ADDRESS**
- **TIME ZONE**

Selectable operating options are:

- **ON**: enables synchronization with the configured NTP server.
- **OFF**: disables NTP synchronization and uses the manually configured date and time.



The **NTP SERVER ADDRESS** identifies the server used as the time reference. The server must be reachable through the configured network connection. **Important:** If a dedicated **NTP server** is available, it can be configured by opening the **NTP SERVER** section, scrolling to the bottom of the page, and selecting the following option: IP NTP Server. A communication with an external NTP server requires valid IP, subnet, gateway and, when a host name is used, DNS settings.

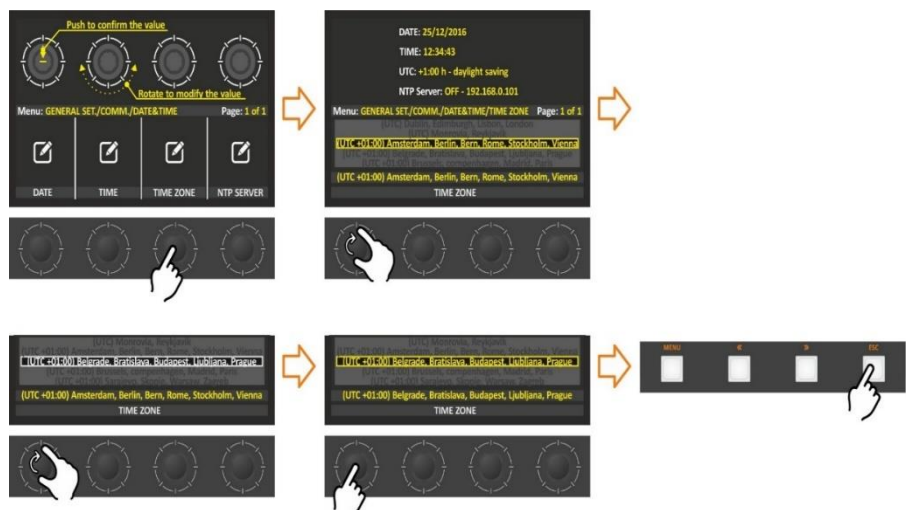


NTP Server in the world:

- **europa.pool.ntp.org** (217.147.223.78)
- **asia.pool.ntp.org** (140.130.175.9)
- **oceania.pool.ntp.org** (203.23.237.200)
- **north-america.pool.ntp.org** (66.250.45.2)
- **south-america.pool.ntp.org** (146.164.53.65)
- **africa.pool.ntp.org** (196.25.1.9)

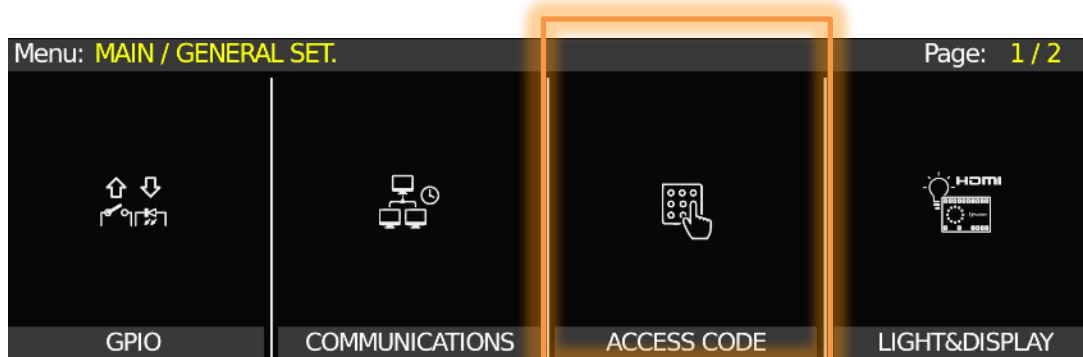
TIME ZONE:

TIME ZONE defines the console's geographical time zone and the corresponding offset from **UTC — Coordinated Universal Time**. The correct time zone must be selected to ensure that the synchronized NTP time is displayed as the correct local time. The selected entry may also include the applicable **daylight-saving-time rules**, depending on the chosen region and firmware implementation.



3. ACCESS CODE

The Oxygen console gives the user the ability to lock the settings with a passcode to prevent any modifications by unauthorized persons. There are two different Access Codes (**4-digit PIN**) to help you if one of the **two passcodes** is forgotten.



- **Two** different Passcode
- The default passcode is **0000**
- It can disable and enable both of them together by **ON/OFF** selection.
- Can set the passcode to lock the display after **10, 30, or 60 minutes**.

ENABLE: If **No** is selected, the console will never be locked. If **Yes** is selected, the console will be locked after the Unlock Time selected in the last parameter of this section. After the Unlock Time, the console needs **CODE1** or **CODE2** to be unlocked successfully.

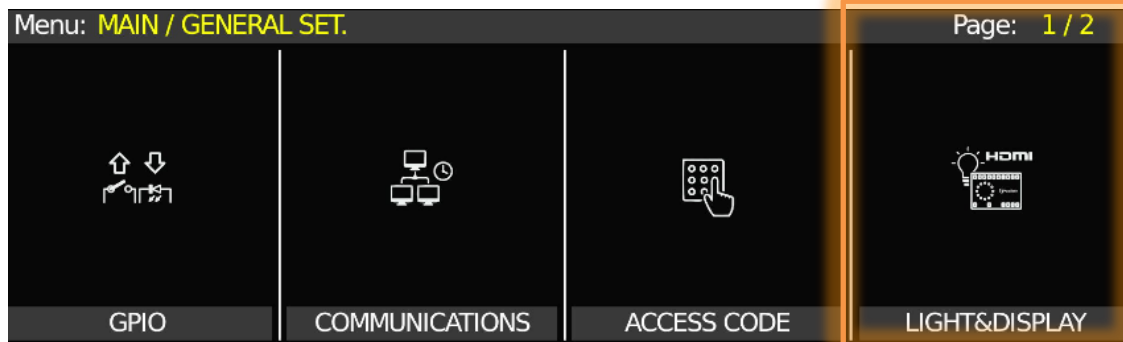
CODE1: here you can set the primary desired code to Unlock the console after Unlock Time will be passed.

CODE2: here you can set the secondary desired code to Unlock the console after Unlock Time will be passed.

UNLOCK TIME: Select between the available options the time after you want the console will be automatically locked. Selectable options are **10 min, 30 min, and 60 min**.

4. LIGHT & DISPLAY

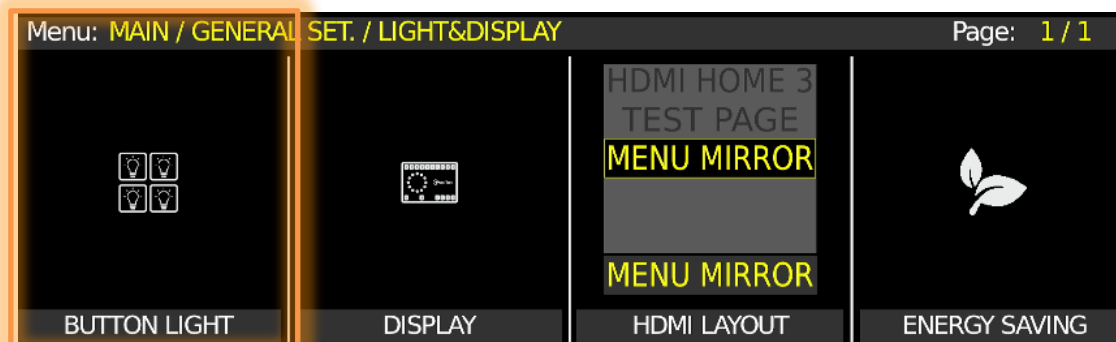
The **LIGHT & DISPLAY** section contains the settings used to configure the console's illuminated controls, integrated display and external HDMI monitor. These parameters improve the visibility of operating information under different studio lighting conditions. They affect the **visual interface only** and do not change audio routing, processing or signal levels.



The section includes:

- **BUTTON LIGHT**
- **DISPLAY**
- **HDMI LAYOUT**
- **ENERGY SAVING**

BUTTON LIGHT:



The **BUTTON LIGHT** menu contains the global settings used to configure the brightness and colour of the illuminated controls on the OXYGEN 3000 Plus surface. These settings help the operator recognise the status and function of each control immediately, particularly in studios with low ambient lighting.

Selectable colour palette

Where a colour-selection parameter is provided, the available list may include the following standard colours:

- **RED**
- **BLUE**

- GREEN
- YELLOW
- CYAN
- MAGENTA
- WHITE

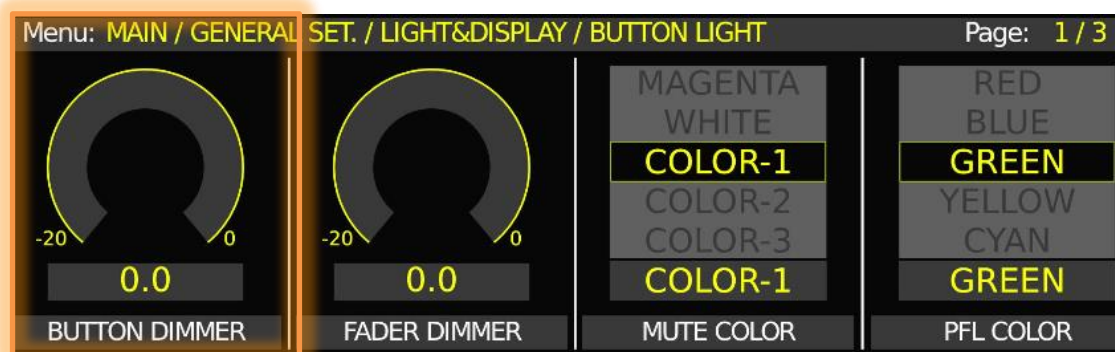
The following user-programmable colour slots may also be available:

- COLOR 1
- COLOR 2
- COLOR 3
- COLOR 4

The exact list displayed by the console may vary according to the **installed firmware version**.

INPUTS SETTING (BUTTON LIGHT)

BUTTON DIMMER:



BUTTON DIMMER adjusts the overall brightness of the illuminated buttons on the console surface.

Increasing the setting makes the button illumination more visible, while decreasing it reduces the brightness for operation in dark studio environments. This parameter normally affects the console buttons globally. It does not change the brightness of the displays or the fader LED bars.

Selectable setting: Adjustable brightness level: select the required intensity using the corresponding encoder. **Range:** -20.0 to 0. **Adjustment step:** 1.0.

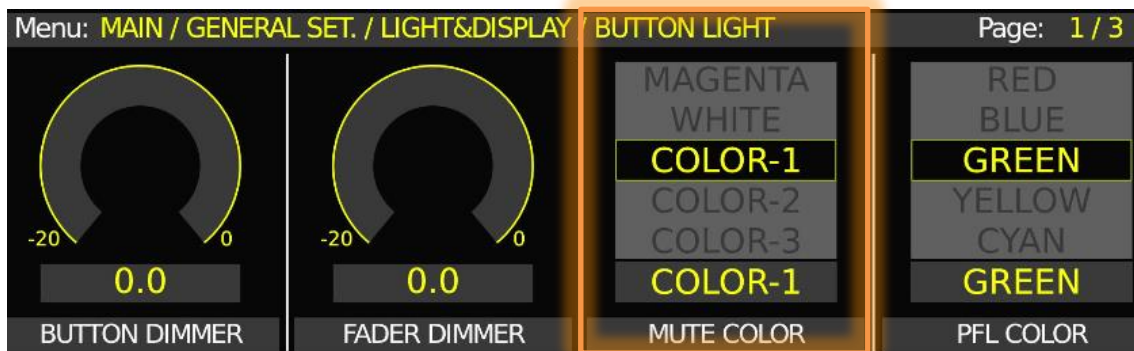
FADER DIMMER:



FADER DIMMER adjusts the brightness of the illuminated elements associated with the faders, particularly the **fader LED bars**. It can be used to match the fader illumination with the brightness of the console buttons and displays.

Selectable setting: Adjustable brightness level: increases or decreases the global intensity of the fader illumination. **Range: -20.0 to 0. Adjustment step: 1.0.**

MUTE COLOR:



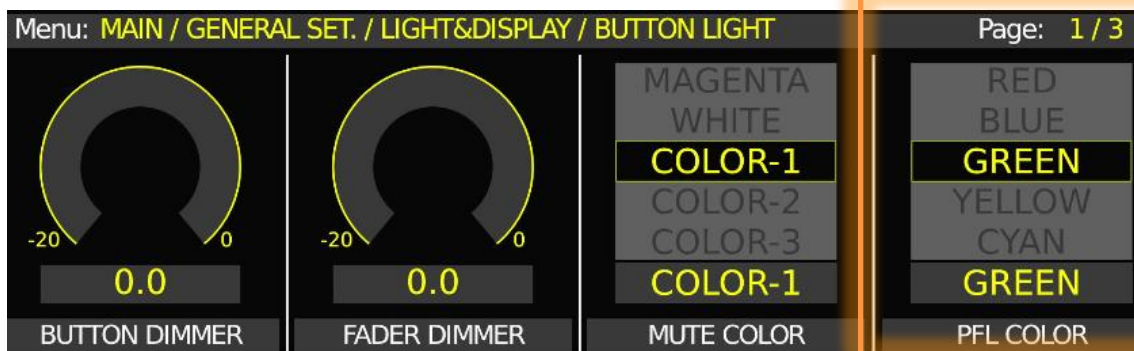
MUTE COLOR selects the colour used to identify a channel in its **muted or standby state**. The selected colour is normally applied to the channel's **ON/START button** when the audio source is not open.

Selectable options:

- One of the available **standard colours**
- **COLOR 1**
- **COLOR 2**
- **COLOR 3**
- **COLOR 4**

This colour provides a visual indication only and does not determine how the channel is activated or muted.

PFL COLOR:



PFL COLOR selects the colour displayed when **PFL — Pre-Fader Listen** is active.

It allows the operator to identify immediately which channel is being monitored before the fader and before the signal is sent to the main output buses. Depending on the firmware configuration, the selected colour may also be used for:

- Active **PGM** buttons
- Active **SUB** buttons
- Active **AUX 1** buttons
- Active **AUX 2** buttons
- The selected output-meter indication in the main monitor section

Selectable options:

- One of the available **standard colours**
- **COLOR 1–COLOR 4**

MENU COLOR:



MENU COLOR selects the illumination colour used for the **MENU button** and the related menu-navigation controls. This setting helps distinguish the configuration controls from the audio-channel and monitor controls. **Selectable options:**

- One of the available **standard colours**
- **COLOR 1–COLOR 4**

The setting affects the control illumination only and does not change the graphical colours of the integrated TFT display.

HDP COLOR:



HDP COLOR selects the colour used for the active **headphone-monitoring controls**. The selected colour identifies the active source-selection button in the **HDP — Headphones** sections, such as the Control Room or Studio headphone monitoring area. **Selectable options:**

- One of the available **standard colours**
- **COLOR 1–COLOR 4**

SPK COLOR:



SPK COLOR selects the colour assigned to the active **speaker-monitoring controls**. It is used to identify the selected monitoring source in the **SPK — Speakers** sections for the Control Room or Studio. **Selectable options:**

- One of the available **standard colours**
- **COLOR 1–COLOR 4**

This parameter does not change the selected monitor source, speaker level or CUT/ATT operating mode.

CUSTOM COLOR:

CUSTOM COLOR provides access to the four user-programmable colour slots:

- **COLOR 1**
- **COLOR 2**
- **COLOR 3**
- **COLOR 4**

Each custom colour is created by adjusting its three **RGB components**:

- **RED:** from **0** to **255**
- **GREEN:** from **0** to **255**
- **BLUE:** from **0** to **255**

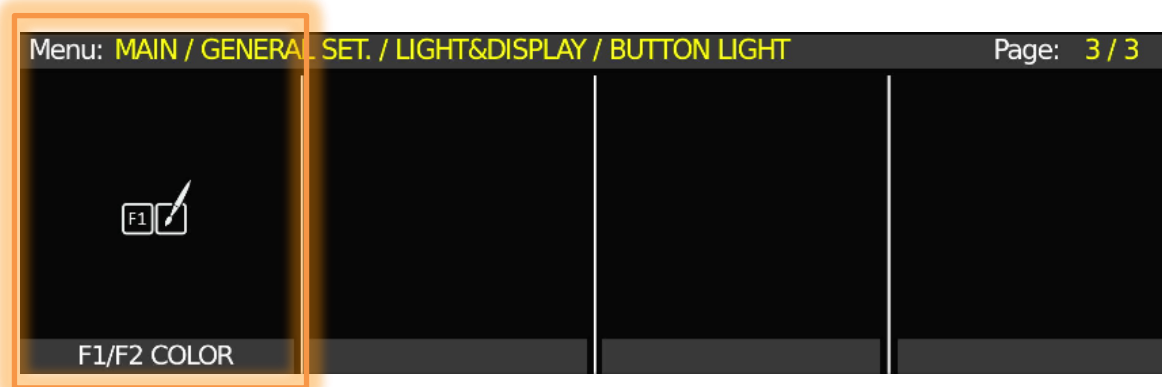
A value of **0** disables the corresponding colour component, while **255** represents its maximum intensity. After a custom colour has been configured, the corresponding **COLOR 1–COLOR 4** entry can be selected in the compatible colour-assignment menus.

R	255	255	255	125	0	0	0	0	125	255	255
G	0	125	255	255	255	255	125	0	0	0	0
B	0	0	0	0	0	125	255	255	255	255	125



F1/F2 COLOR:

F1/F2 COLOR selects the illumination colour used by the channel's programmable **F1 and F2 function buttons**. The selected colour provides visual feedback when the function assigned to F1 or F2 is active.



Depending on the selected input source, these buttons may control functions such as:

- EQ
- COMPRESSOR
- NOISE GATE
- DUCKING
- TALKBACK
- Telephone or TELCO control functions

Selectable options:

- One of the available **standard colours**
- **COLOR 1–COLOR 4**

Important: F1/F2 COLOR does not assign a function to the buttons. The F1 and F2 functions must be configured separately in the input source's **F1 MODE** and **F2 MODE** settings. The meaning of the illuminated state therefore depends on the function assigned to each button.



DISPLAY:

The **DISPLAY** menu contains the settings used to configure the brightness and visual content of the displays installed on the OXYGEN 3000 Plus. It provides separate controls for:

- The central **7-inch TFT menu display**
- The Home-page graphical layout
- The customised station logo
- The individual **channel displays** located above the faders

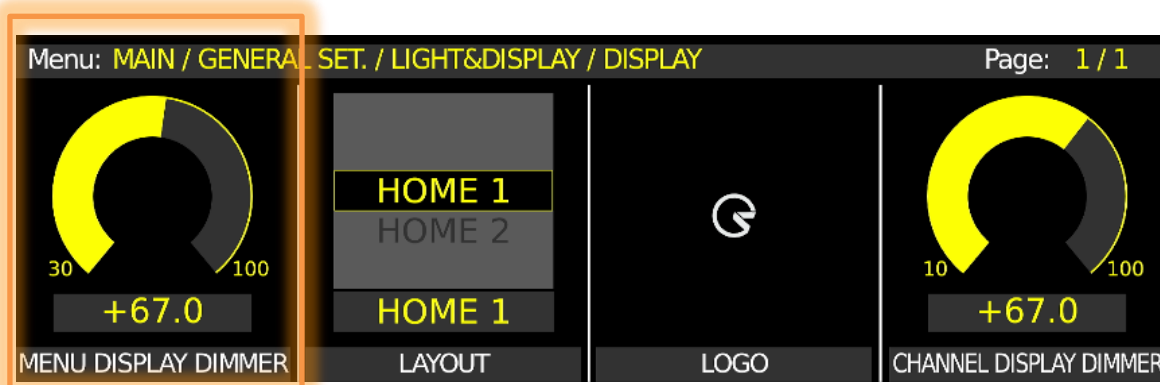


The menu includes:

- **MENU DISPLAY DIMMER**
- **LAYOUT**
- **LOGO**
- **CHANNEL DISPLAY DIMMER**

INPUTS SETTING (DISPLAY)

MENU DISPLAY DIMMER:



MENU DISPLAY DIMMER adjusts the brightness of the console's central **7-inch, 800 × 480 pixel TFT display**. It allows the display illumination to be adapted to different studio conditions. A lower brightness is useful in dark studios, while a higher brightness improves visibility in brightly illuminated environments. **Range: 30.0 to 100.0. Adjustment step: 1.0**

LAYOUT:

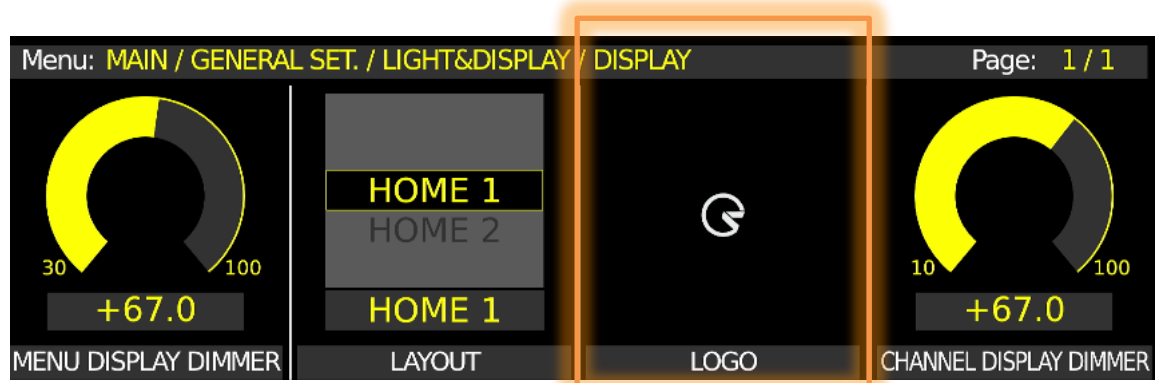


LAYOUT selects the graphical arrangement used on the central display when the console is on its normal **Home page** and no configuration menu is open.

The documented options are:

- **HOME 1:** displays the standard Home-page layout, including the console clock, station logo, channel information and operating-status indications.
- **HOME 2:** displays an alternative layout that also includes the **Control Room and Studio microphone counters**, together with the clock and station logo.

LOGO:



LOGO allows a customised radio-station logo to be imported and displayed on the console's Home page.

The selectable commands are:

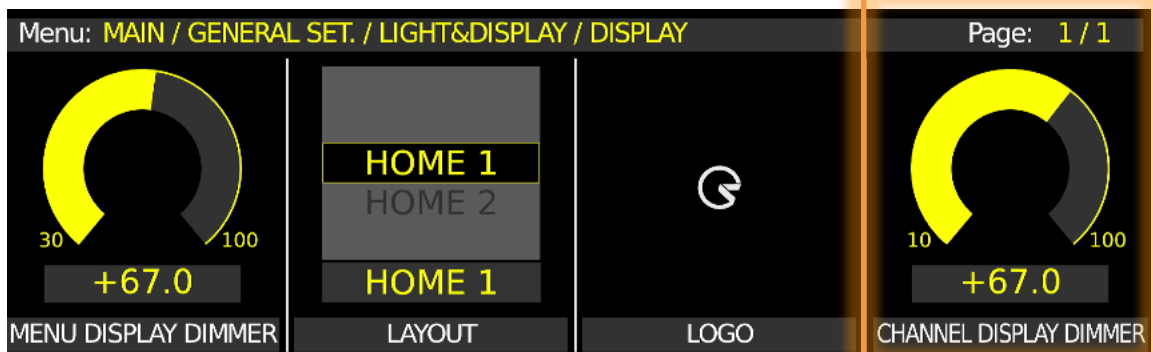
- **READ USB:** imports a customised logo from a USB memory device.
- **RESTORE DEFAULT:** removes the customised logo and restores the factory image.

To import a customised logo:

1. Prepare an image with a resolution of exactly **330 × 280 pixels**.
2. Save the image in **PNG format**.
3. Name the file exactly **logo.png**.
4. Copy the file to a USB memory device.
5. Connect the USB device to one of the console's rear USB ports.
6. Select **READ USB** and confirm the operation.

If the console does not recognise the file, verify its **filename, dimensions and format**.

CHANNEL DISPLAY DIMMER:



CHANNEL DISPLAY DIMMER adjusts the brightness of the individual TFT displays located above the console's physical faders. The setting is normally applied globally to all compatible channel displays, ensuring consistent brightness across the console surface. **Range: 30.0 to 100.0. Adjustment step: 1.0**

HDMI LAYOUT:

The **HDMI LAYOUT** menu selects the graphical interface sent by the OXYGEN 3000 Plus to an external monitor connected to the console's rear **HDMI output**. The external display provides a larger, continuously visible overview of the console status. Depending on the selected layout, it can show channel assignments, signal levels, timers, monitor status and other operational information.



The selectable options are:

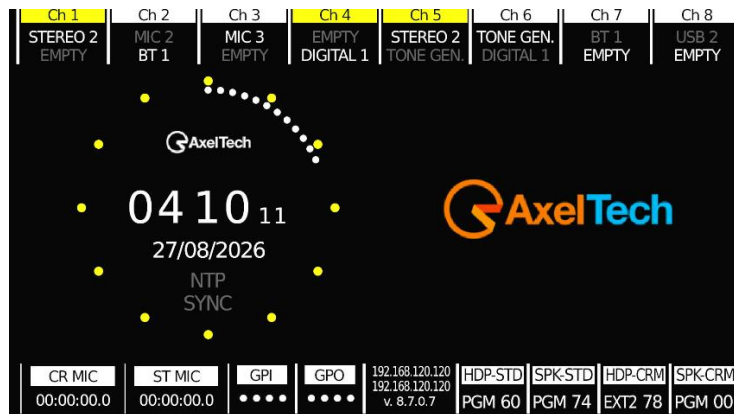
- **HDMI HOME 1**
- **HDMI HOME 2 16 CH**
- **HDMI HOME 2 24 CH**
- **HDMI HOME 3**
- **TEST PAGE**
- **MENU MIRROR**

Only one HDMI layout can be active at a time.

HDMI HOME 1:

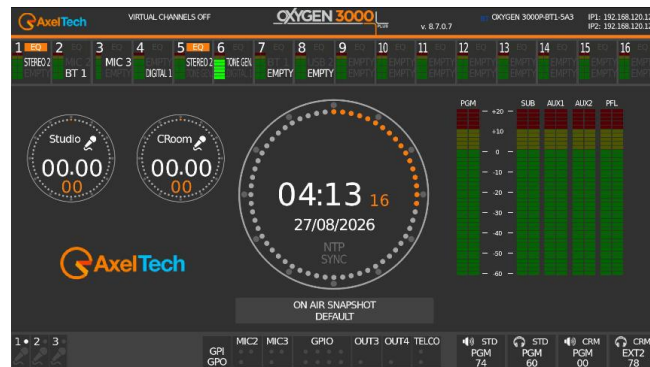
HDMI HOME 1 selects the first standard operating layout for the external HDMI monitor.

This layout is suitable when a complete overview of both channel and system information is required.



HDMI HOME 2 16 CH:

HDMI HOME 2 16 CH selects a version of the second HDMI layout designed to display up to **16 channels**. It provides a channel-oriented representation with sufficient space for clearly displaying the main information associated with each channel. This layout is recommended for consoles configured with up to **16 active physical or logical channels**. If fewer than 16 channels are configured, the unused positions may remain empty or inactive, depending on the firmware version.



HDMI HOME 2 24 CH:

HDMI HOME 2 24 CH expands the second HDMI layout to display up to **24 channels** simultaneously.

This option is particularly useful when the console uses the maximum configuration of:

- Up to **16 physical channels**
- Up to **8 additional virtual channels**

The virtual channels are configured and controlled through **OXYGEN REMOTER**. When they are enabled, the 24-channel layout allows physical and virtual channels to be monitored together on the same external screen. Because more channels are shown simultaneously, the individual channel fields are more compact than in the 16-channel version.

Important: selecting HDMI HOME 2 24 CH does not automatically enable the virtual channels. They must be enabled separately in the console's channel settings and configured through OXYGEN REMOTER.

HDMI HOME 3:

HDMI HOME 3 selects the third factory-defined HDMI operating layout.

It provides an alternative arrangement of channel, monitoring and system-status information. It can be selected when its graphical organisation is more suitable for the studio's operating requirements or external-monitor dimensions.



TEST PAGE:

TEST PAGE displays a diagnostic image on the external HDMI monitor. It can be used during installation or maintenance. TEST PAGE does not show the live operating status of the mixer and is therefore **not intended for normal On-Air operation**. After completing the test, select one of the HDMI HOME layouts or MENU MIRROR to return to the normal console display.

MENU MIRROR:

MENU MIRROR reproduces the content currently displayed on the console's integrated **7-inch menu display** on the external HDMI monitor. This mode allows the operator or technical staff to view the console menus on a larger screen. The external monitor acts as a **visual replica only**. Menu navigation and parameter adjustment must still be performed using the console's buttons and rotary encoders or through an authorised remote-control interface. **Important:** MENU MIRROR displays the current menu page instead of the normal HDMI status overview. To restore permanent channel and meter monitoring, select one of the HDMI HOME layouts.

ENERGY SAVING:

The **ENERGY SAVINGS** menu configures the automatic reduction of the console's display and control illumination after a specified period of operator inactivity. When the configured timeout expires, the OXYGEN 3000 Plus applies the brightness levels defined in this menu to the buttons, fader indicators and displays. When operator activity resumes, the interface normally returns to its standard operating brightness.



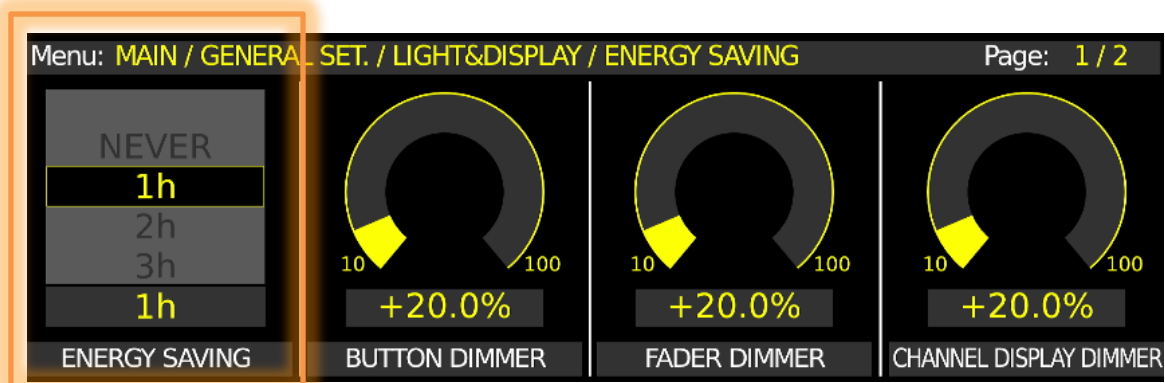
The menu includes:

- **ENERGY SAVINGS**
- **BUTTON DIMMER**
- **FADER DIMMER**
- **CHANNEL DISPLAY DIMMER**
- **MENU DISPLAY DIMMER**

Important: the Energy Savings function affects the console's illumination and displays only. It does not switch off the console, mute any channel or interrupt audio processing, routing, network connections or On-Air operation.

INPUTS SETTING (DISPLAY)

ENERGY SAVINGS:



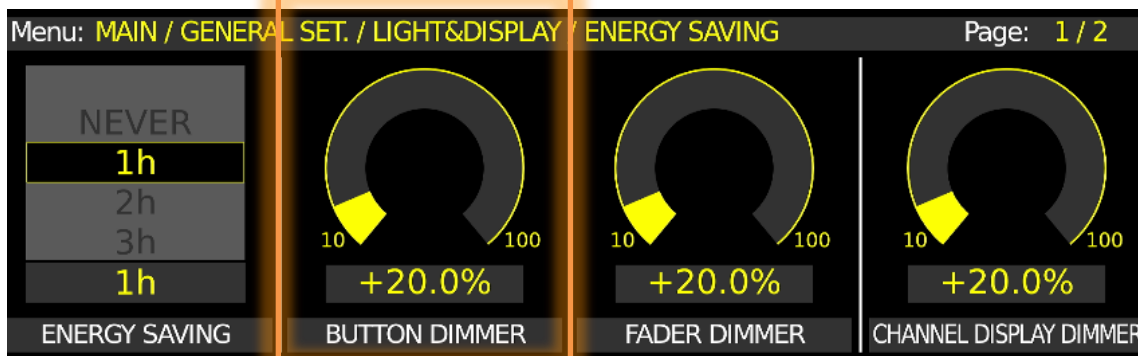
ENERGY SAVINGS defines the period of inactivity after which the console enters its energy-saving state.

The selectable options are:

- **NEVER:** disables automatic activation of the energy-saving state.
- **1h:** activates Energy Savings after **1 hour** of inactivity.
- **2h:** activates Energy Savings after **2 hours** of inactivity.
- **3h:** activates Energy Savings after **3 hours** of inactivity.
- **4h:** activates Energy Savings after **4 hours** of inactivity.

When **NEVER** is selected, the energy-saving dimmer values remain stored but are not applied automatically. When a timed option is selected, interaction with the console controls normally restores the standard lighting levels and restarts the inactivity timer.

BUTTON DIMMER:



BUTTON DIMMER defines the brightness level applied to the console's illuminated buttons when Energy Savings is active. **Range: 10.0 % to 100%. Adjustment range: 10.0%**

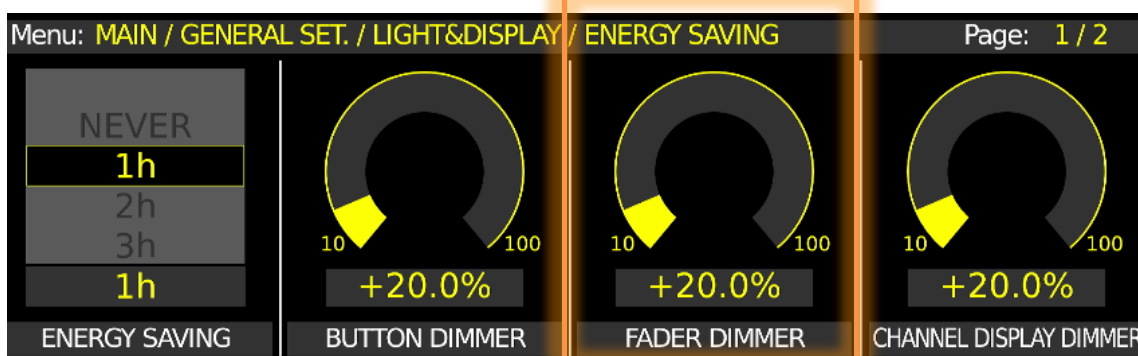
It can reduce the illumination of controls such as:

- ON/START buttons
- PFL buttons
- Bus-assignment buttons
- Menu-navigation buttons
- Monitor-selection buttons
- F1 and F2 function buttons

This value is independent of the normal **BUTTON DIMMER** setting available in the BUTTON LIGHT menu.

Important: BUTTON DIMMER changes the brightness only. Button colours and operating states remain unchanged.

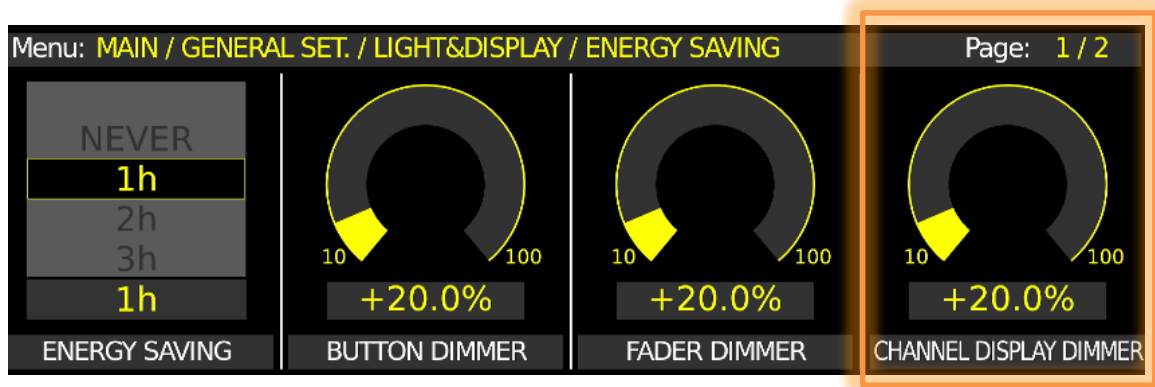
FADER DIMMER:



FADER DIMMER defines the brightness applied to the illuminated elements associated with the faders during the energy-saving state. It mainly affects the **fader LED bars** and other compatible fader illumination. A lower illumination level reduces unnecessary light output while retaining sufficient visual feedback, if required. **Range: 10.0 % to 100%. Adjustment range: 10.0%**

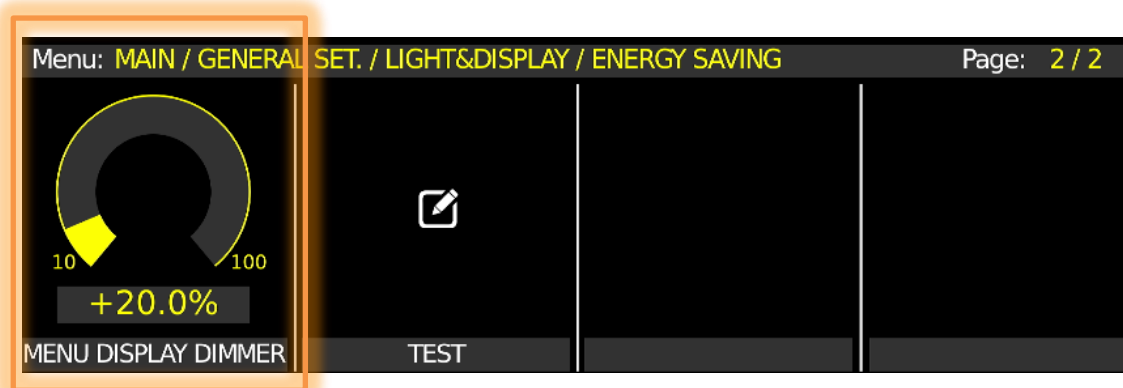
Important: this parameter does not move the faders and does not change their audio attenuation. It affects illumination only.

CHANNEL DISPLAY DIMMER:



CHANNEL DISPLAY DIMMER defines the brightness of the individual channel displays when Energy Savings is active. These are the small TFT displays located above the physical faders. **Range: 10.0 % to 100%. Adjustment range: 10.0%**

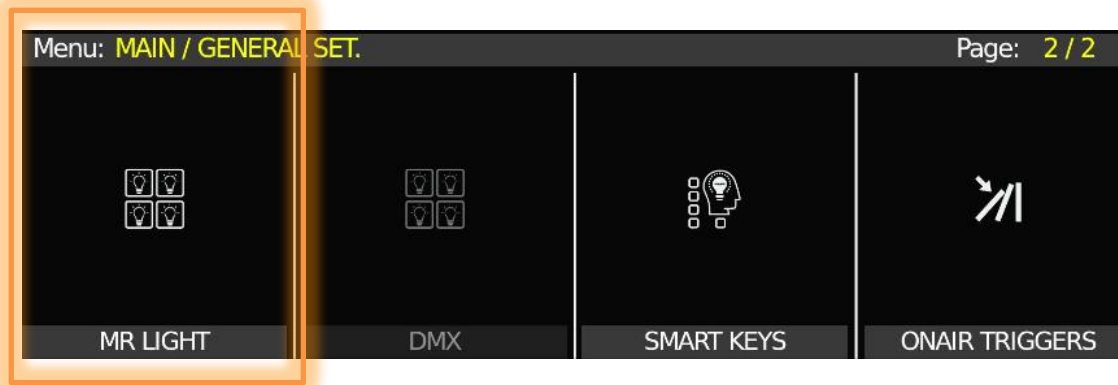
MENU DISPLAY DIMMER



MENU DISPLAY DIMMER defines the brightness of the central 7-inch TFT menu display during the energy-saving state. **Range: 10.0 % to 100%. Adjustment range: 10.0%.** This parameter allows the central display to be dimmed independently from the buttons, fader indicators and channel displays. It does not change the selected Home-page layout, station logo or HDMI output configuration.

5. MR LIGHT

The **MR LIGHT** menu configures the integration between the OXYGEN 3000 Plus and compatible **Axel Technology MR. LIGHT RGB** signalling lamps.



MR. LIGHT RGB is a multipurpose studio lamp designed to provide clear visual indications in radio, television, production and recording environments. Depending on its configuration, it can display different **colours and lighting patterns** to indicate conditions such as:

- **On Air**
- **Microphone open**
- **Studio active**
- **Control Room active**

The supported OXYGEN firmware provides **automatic device discovery** and integration with the console's **On Air Light events**. Compatible network-enabled MR. LIGHT units can be controlled through the studio network, including the supported Wi-Fi connection. The MR LIGHT menu contains the following logical device positions:

- **MRL-CR1 and MRL-CR2**
- **MRL-ST1 and MRL-ST2**
- **MRL-1 through MRL-8**

Important: these entries are **logical lamp assignments**, not audio sources or audio-processing functions. They determine which physical MR. LIGHT device receives the events associated with a specific room or general-purpose lighting position.

MRL-CR1 / MRL-CR2:

The **MRL-CR1** and **MRL-CR2** menus configure the two logical MR. LIGHT RGB devices associated with the **Control Room**.

- **MRL-CR1:** first Control Room signalling lamp
- **MRL-CR2:** second Control Room signalling lamp

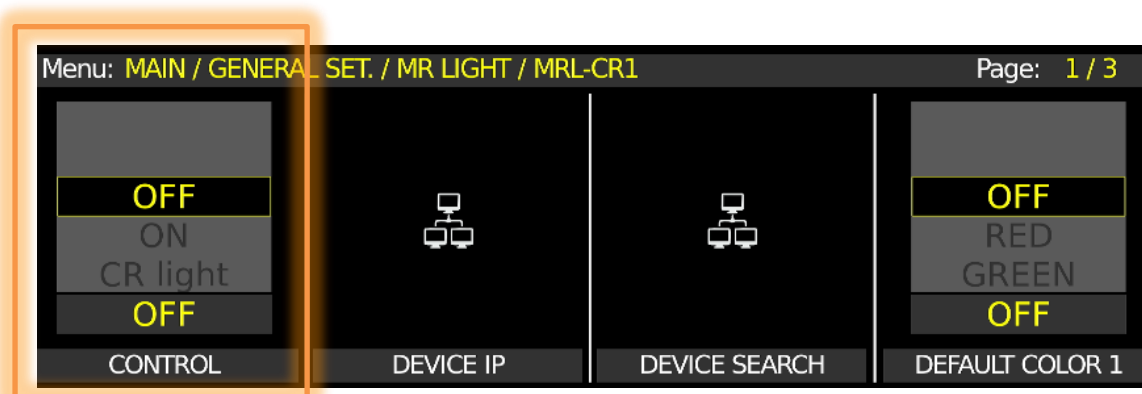


The two devices are configured independently. They can be used, for example, to provide the same indication in two different positions or to display separate operational states inside and outside the Control Room. MR. LIGHT RGB supports multicolour illumination and different lighting effects, including fixed, flashing, two-colour and pulsing indications. The MRL-CR1 and MRL-CR2 configuration pages contain:

- **CONTROL**
- **DEVICE IP**
- **DEVICE SEARCH**
- **DEFAULT COLOR 1**
- **DEFAULT COLOR 2**
- **DEFAULT EFFECT**
- **COLOR OFF 1**
- **COLOR OFF 2**
- **EFFECT OFF**

INPUTS SETTING (MRL-CR1 e MRL-CR2)

CONTROL:



CONTROL defines how the selected MR. LIGHT device is managed by the OXYGEN 3000 Plus and which logical lighting group it follows. The selectable options are:

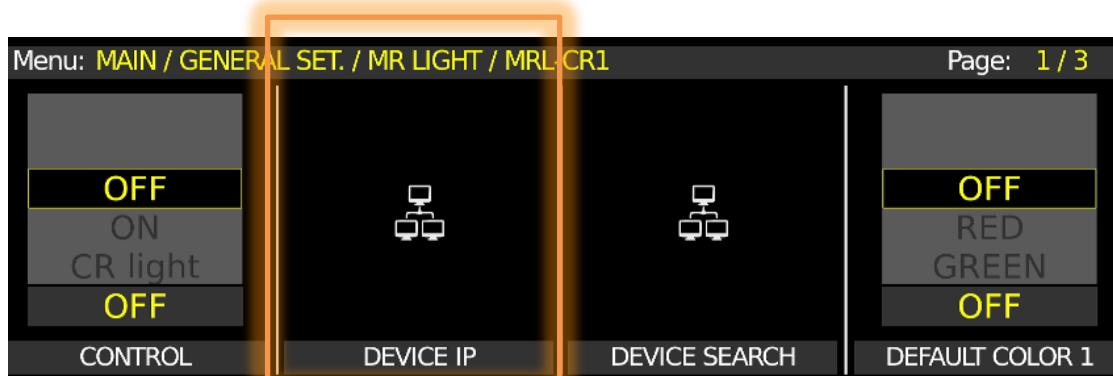
- **ON:** enables direct control of the lamp according to the configured parameters of the selected MRL position.
- **OFF:** disables control of the selected lamp from the console.

- **CR LIGHT:** links the lamp to the **Control Room lighting logic**, allowing it to follow Control Room On Air and microphone-related events.
- **ST LIGHT:** links the lamp to the **Studio lighting logic**, allowing it to follow Studio On Air and microphone-related events.

When **CONTROL** is set to **OFF**, the device configuration may remain stored, but the console does not send operational lighting commands to that MRL position.

When **CR LIGHT** or **ST LIGHT** is selected, the lamp behaviour is driven by the corresponding system-wide lighting events rather than only by the local MRL configuration. **Important:** **CONTROL** affects only the selected logical device—**MRL-CR1** or **MRL-CR2**—and does not disable or modify other MR. LIGHT units configured in the system.

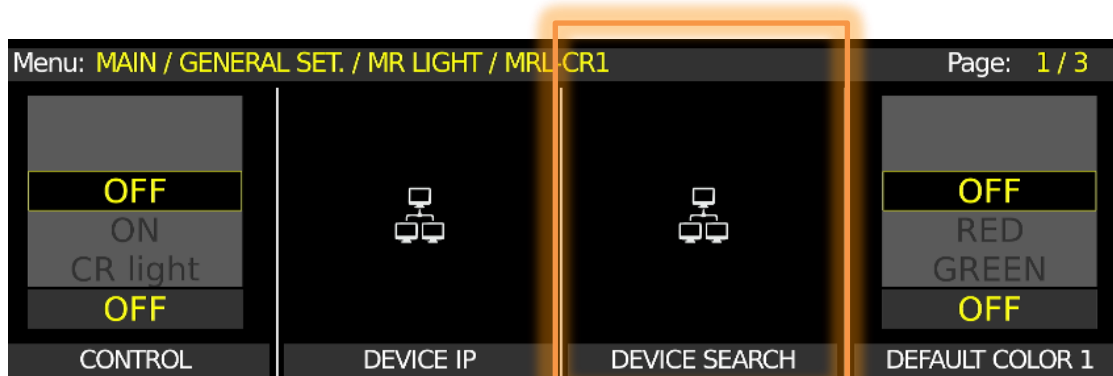
DEVICE IP:



DEVICE IP specifies the network address of the physical MR. LIGHT device assigned to the selected logical position. The address identifies the lamp to which the OXYGEN console must send its control commands. The console and the lamp must be connected to a **reachable network**, normally using compatible IP-address and subnet settings. Each physical MR. LIGHT device must have its own network identity. Assigning an incorrect or unreachable address prevents the console from controlling the lamp.

Important: **DEVICE IP** is a network-control parameter. It does not carry audio and does not modify the console's audio routing. Moreover you have to set manually the IP.

DEVICE SEARCH:



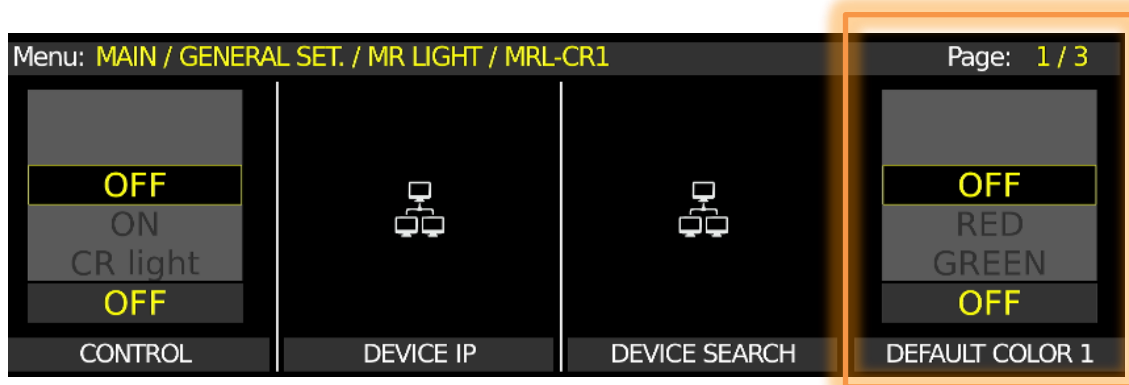
DEVICE SEARCH starts an automatic search for compatible MR. LIGHT devices available on the network.

This function can be used to:

- Detect compatible lamps without manually entering their address
- Verify that a device is reachable
- Identify the address of a newly installed lamp
- Associate a detected device with **MRL-CR1** or **MRL-CR2**

The OXYGEN firmware supports **automatic device discovery** for MR. LIGHT integration. **DEVICE SEARCH** is an **execution command**, not a permanent operating mode. After selecting a device, verify that the correct address appears under **DEVICE IP**.

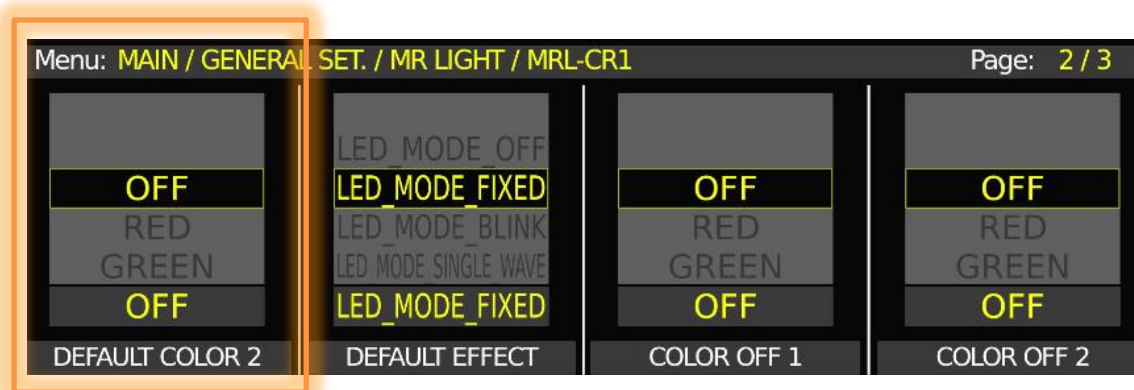
DEFAULT COLOR 1:



DEFAULT COLOR 1 selects the primary RGB colour used for the lamp's default operating indication. This setting defines the lamp color displayed when the microphone is set to **ON**. **Selectable options:**

- One of the available **standard colours**
- **COLOR 1–COLOR 4**
- **DO NOT SET IT OFF**

DEFAULT COLOR 2:



DEFAULT COLOR 2 selects the secondary RGB colour used by two-colour lighting effects. This setting defines the lamp color displayed when the microphone is set to **ON**. **Selectable options:**

- One of the available **standard colours**
- **COLOR 1–COLOR 4**
- **OFF**: disables the use of a second colour, forcing single-colour behaviour even when a two-colour effect is selected

COLOR 2 is used together with **DEFAULT COLOR 1** when **DEFAULT EFFECT** is configured for a two-colour pattern, such as alternating or pulsing between two colours. If **OFF** is selected, the system ignores the second colour and the effect operates only with **DEFAULT COLOR 1**.

DEFAULT EFFECT:



DEFAULT EFFECT defines the lighting behaviour applied to the default state of the MR. LIGHT device.

The available modes are:

- **LED_MODE_OFF**: turns the LED output completely off
- **LED_MODE_FIXED**: displays a fixed colour without animation
- **LED_MODE_BLINK**: simple on/off flashing of the selected colour
- **LED_MODE_SINGLE_WAVE**: single-direction wave animation using **COLOR 1**
- **LED_MODE_DOUBLE_WAVE**: bidirectional wave animation using **COLOR 1** and **COLOR 2**
- **LED_MODE_FADING_BLINK**: blinking with smooth fade in/out transitions
- **LED_MODE_CIRCLE_WAVE**: circular or rotating wave effect
- **LED_MODE_NOISE**: randomised colour/intensity variation effect
- **LED_MODE_RAINBOW**: continuous full-spectrum colour cycling
- **LED_MODE_RAINBOW_WAVE**: wave-based rainbow animation
- **LED_MODE_RAINBOW_2**: alternative rainbow pattern with different transition logic
- **LED_MODE_KITT**: scanning light effect similar to a moving bar
- **LED_MODE_CIRCLE**: circular motion effect with continuous rotation

Two-colour or advanced effects require both **DEFAULT COLOR 1** and **DEFAULT COLOR 2** to be properly configured, except when the selected mode uses only a single colour. **Important:** the exact behaviour of each mode may vary slightly depending on MR. LIGHT firmware version.

COLOR OFF 1:



COLOR OFF 1 selects the primary colour used when the Control Room lighting condition is in its **inactive** state. **Selectable setting:**

- **One colour from the available RGB palette**
- **OFF:** disables illumination for the OFF state (lamp remains completely dark)

This parameter defines the standby appearance of the lamp when no active Control Room event is present. The inclusion of the **OFF option** allows the operator to force a fully extinguished state instead of a coloured standby indication.

COLOR OFF 2:



COLOR OFF 2 defines the secondary colour used in OFF-state multi-colour effects. **Selectable setting:**

- **One colour from the available RGB palette**
- **OFF:** disables the second colour in OFF-state behaviour

It is only used when the OFF effect supports two-colour or animated behaviour. Otherwise, it has no visible impact.

EFFECT OFF:



EFFECT OFF defines the lighting behaviour of the MR. LIGHT device when the Control Room event is inactive. The available modes typically include:

- **LED_MODE_OFF:** completely disables illumination
- **LED_MODE_FIXED:** displays a static standby colour
- **LED_MODE_BLINK:** flashing standby indication
- **LED_MODE_SINGLE_WAVE:** wave animation using COLOR OFF 1
- **LED_MODE_DOUBLE_WAVE:** wave animation using COLOR OFF 1 and COLOR OFF 2
- **LED_MODE_FADING_BLINK:** smooth fade blinking
- **LED_MODE_CIRCLE_WAVE:** circular wave motion
- **LED_MODE_NOISE:** randomised standby effect
- **LED_MODE_RAINBOW:** full-spectrum cycling in standby
- **LED_MODE_RAINBOW_WAVE:** wave-based rainbow standby animation
- **LED_MODE_RAINBOW_2:** alternative rainbow standby pattern
- **LED_MODE_KITT:** scanning standby effect
- **LED_MODE_CIRCLE:** continuous circular motion

This parameter allows the definition of a clearly recognisable **standby or inactive visual state**, rather than a simple OFF condition.

MRL-ST1 / MRL-ST2:

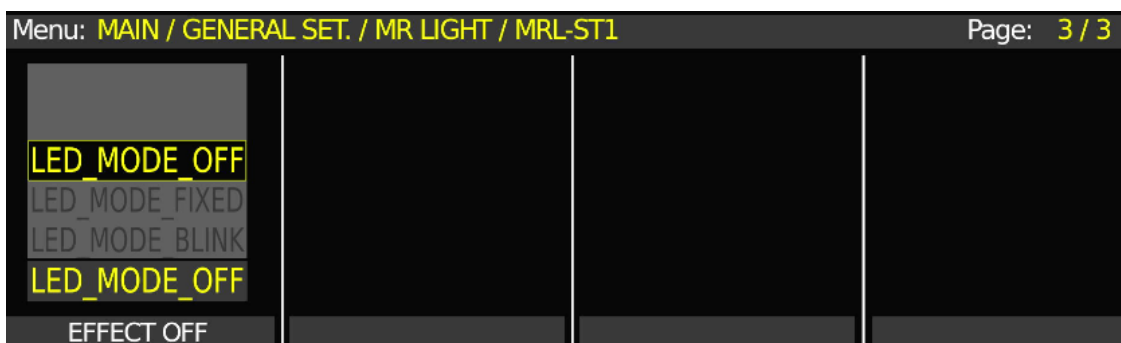
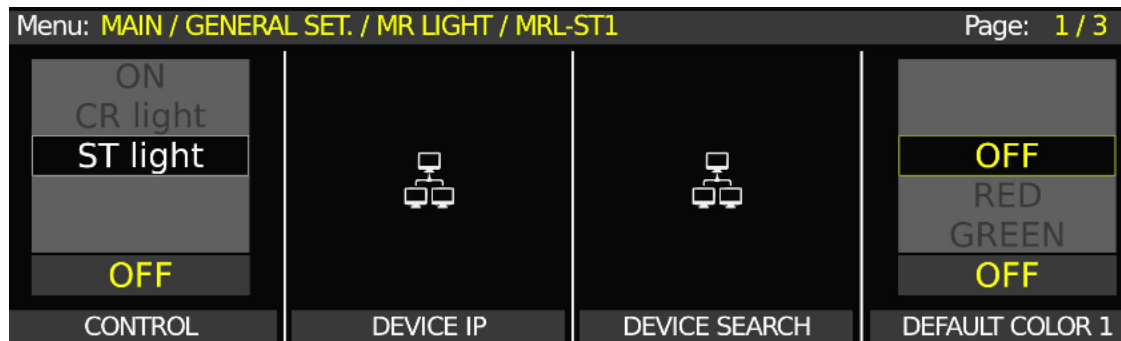
The **MRL-ST1** and **MRL-ST2** menus configure the two logical MR. LIGHT RGB devices associated with the **Studio**.

- **MRL-ST1:** first Studio signalling lamp
- **MRL-ST2:** second Studio signalling lamp



The two devices can be configured independently. They may be installed inside the Studio, at its entrance or in another position where the Studio operating status must be clearly visible. MR. LIGHT RGB supports multicolour illumination and several fixed, flashing, wave-based and animated effects.

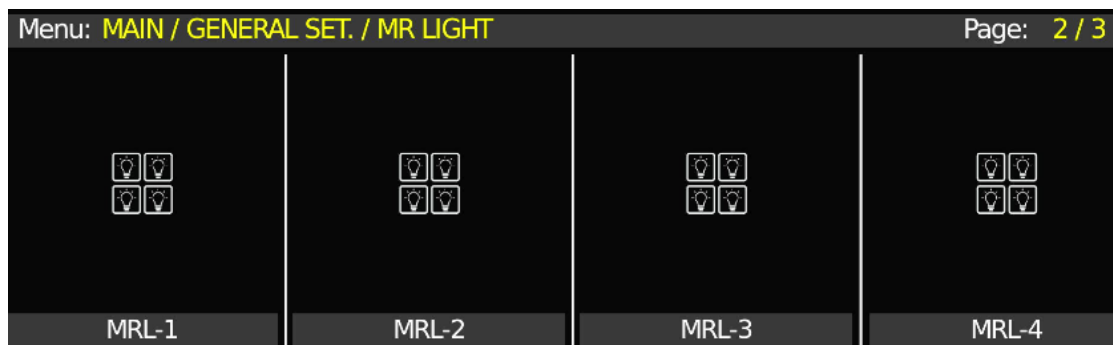
These are the editable options in this section:



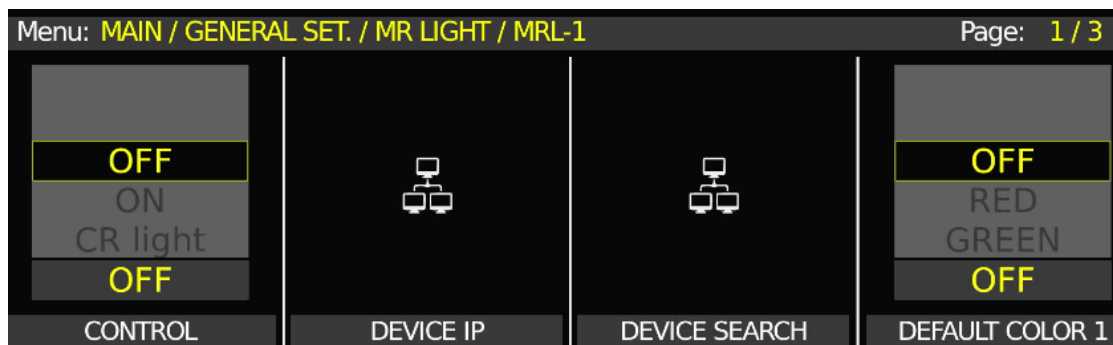
On the **Oxygen 3000 Plus**, the options available in the **MRL-ST1/2** section are identical to those previously described for **MRL-CR1/2**.

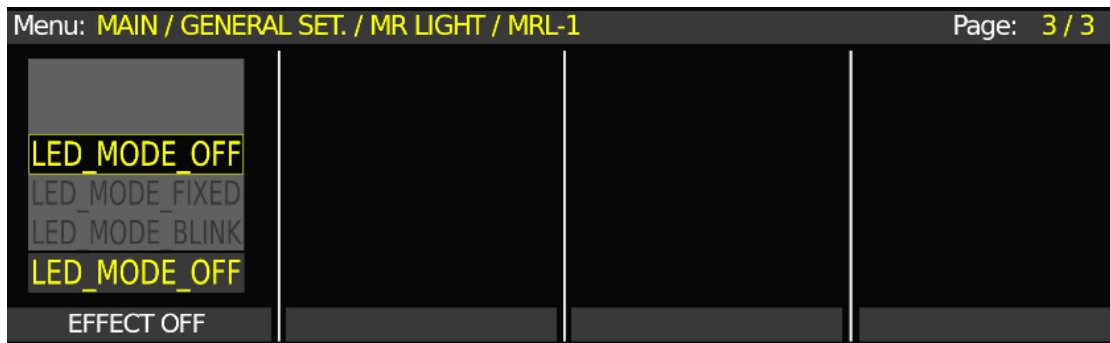
MRL 1-8:

The **MRL 1–8** section provides eight additional logical positions for controlling compatible **MR. LIGHT RGB devices** from the OXYGEN 3000 Plus. Unlike **MRL-CR1/2** and **MRL-ST1/2**, which are specifically identified as Control Room and Studio positions, MRL 1–8 are **general-purpose assignments**. They can therefore be used for additional lamps that are not permanently associated with the main Control Room or Studio positions. Each MRL position can be associated with a separate physical MR. LIGHT device and configured independently.



Each MRL number represents a **logical control position** inside the console. It is not a physical audio input, an audio output or a lighting effect. A physical MR. LIGHT device is associated with the selected position through its **DEVICE IP** or by using **DEVICE SEARCH**. **Important:** each physical MR. LIGHT device must have a valid and unique network address. Assigning an incorrect or duplicated address may prevent reliable lamp control. **These are the editable options in this section:**





On the **Oxygen 3000 Plus**, the options available in the **MRL 1-8** section are identical to those previously described for **MRL-CR1/2**.

6. DMX

The **DMX** menu, located under **GENERAL SETTINGS**, configures the integration between the OXYGEN 3000 Plus and a compatible **DMX lighting-control system**. It can be accessed through:

MAIN → GENERAL SETTINGS → DMX

DMX is a digital control protocol commonly used to manage professional lighting equipment. The OXYGEN 3000 Plus uses DMX control to associate lighting operations with console events, such as opening a Control Room or Studio microphone

The DMX section includes:

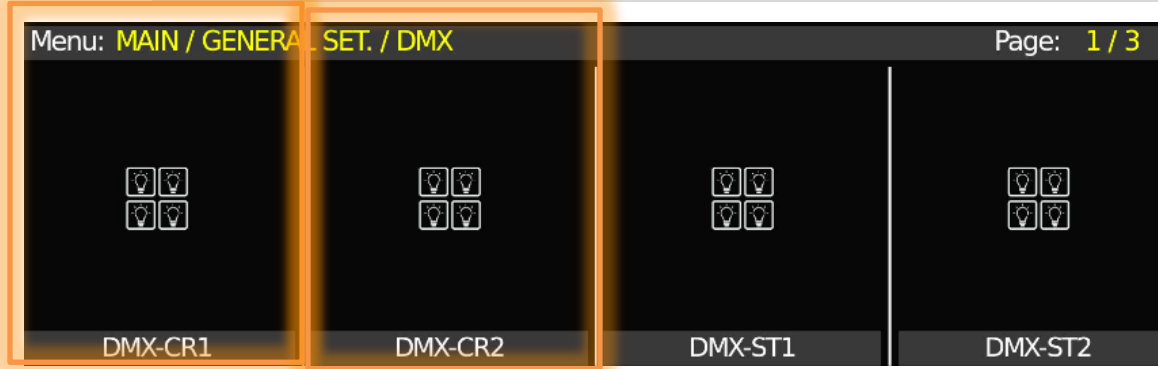
- **DMX-CR1**
- **DMX-CR2**
- **DMX-ST1**
- **DMX-ST2**
- **DMX 1-8**

Important: these entries identify logical DMX control positions. They do not necessarily correspond to twelve separate physical DMX connectors or universes.

DMX-CR1 / DMX-CR2:

The **DMX-CR1** and **DMX-CR2** menus configure the two logical DMX lighting devices associated with the **Control Room**.

- **DMX-CR1:** first Control Room DMX device
- **DMX-CR2:** second Control Room DMX device



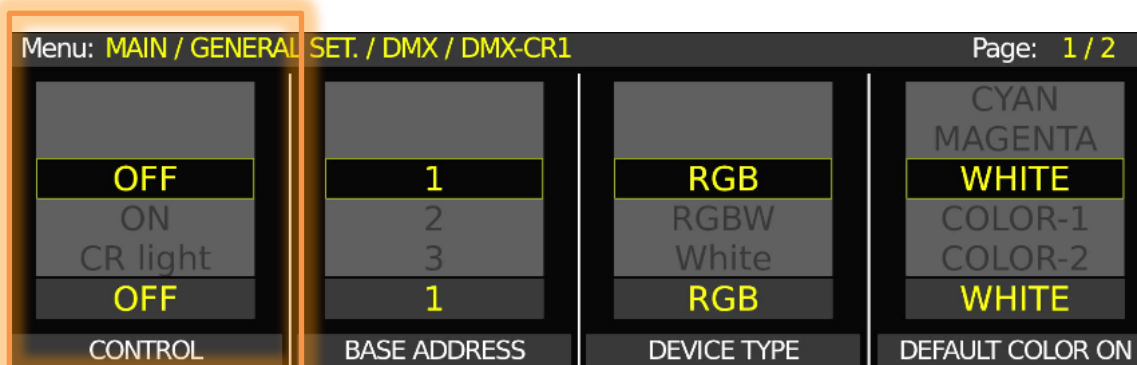
The two positions are configured independently. They can control two separate lighting fixtures, two lighting zones or different attributes of the same DMX installation.

The DMX-CR1 and DMX-CR2 pages contain:

- **CONTROL**
- **BASE ADDRESS**
- **DEVICE TYPE**
- **DEFAULT COLOR ON**
- **DEFAULT COLOR OFF**
- **DEFAULT DIMMER ON**
- **DEFAULT DIMMER OFF**
- **AUTOREFRESH**

INPUT SETTINGS (DMX CR-1/DMX CR-2)

CONTROL:

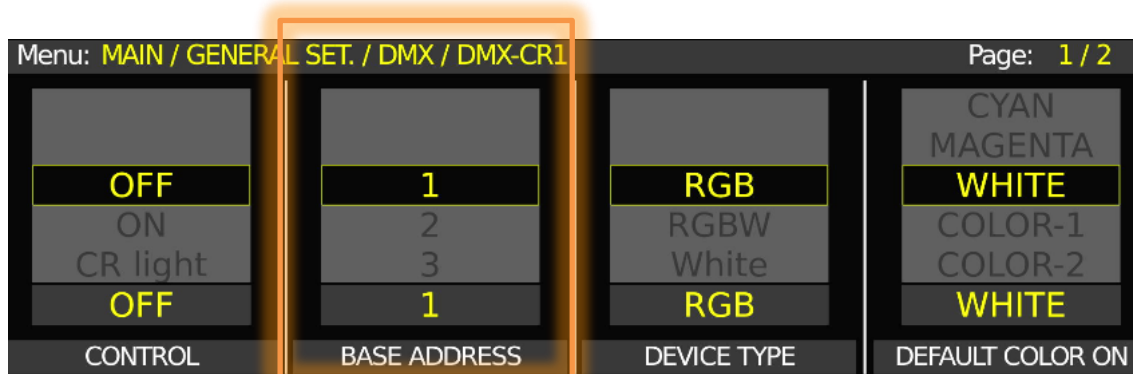


CONTROL enables or disables the selected DMX device and determines which logical lighting event it follows. The selectable options may include:

- **ON:** The selected DMX position is enabled and transmits the configured lighting values according to its assigned operating logic.
- **OFF:** The DMX position is disabled. The address, device type, colours and dimmer values may remain stored, but the console does not use that logical position to control the connected fixture.
- **CR LIGHT:** The DMX device follows the **Control Room lighting logic**. When the Control Room condition becomes active, the console applies: **DEFAULT COLOR ON** and **DEFAULT DIMMER ON**. When the condition becomes inactive, it applies: **DEFAULT COLOR OFF** and **DEFAULT DIMMER OFF**. This is the normal assignment for **DMX-CR1** and **DMX-CR2**.

- **ST LIGHT:** The DMX device follows the **Studio lighting logic**. This selection can be used when a device configured from the DMX-CR page must respond to Studio events instead of Control Room events.
- **ANY TEL/BT RING:** The DMX device is activated when **any telephone or Bluetooth call ring event** is detected by the system. When the ring event is active, the console applies the **ON state parameters** (COLOR ON and DIMMER ON). When the ring event ends, it returns to the **OFF state parameters**.
- **FIND ME:** This mode is typically used for identification and maintenance purposes, allowing a connected fixture to be easily located in a complex installation. When FIND ME is active, the console forces a visible lighting state (for example maximum brightness or a distinctive colour pattern depending on configuration). When FIND ME is deactivated, the device returns to its normal operating logic.

BASE ADDRESS:



BASE ADDRESS specifies the first DMX channel used by the selected lighting device. **Selectable setting:**

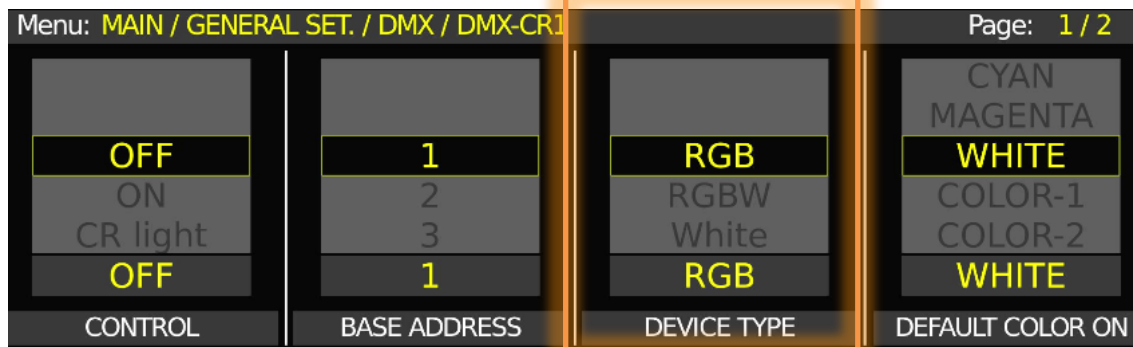
- A valid DMX base address within the supported universe, normally **1 to 508**

A fixture that uses multiple parameters occupies several consecutive channels beginning with the selected BASE ADDRESS. For example, if an RGB device uses three channels and BASE ADDRESS is set to **100**, the console may use:

- **Channel 100:** Red
- **Channel 101:** Green
- **Channel 102:** Blue

A fixture with a separate master dimmer may require an additional channel. The exact channel order depends on the selected **DEVICE TYPE** and on the fixture's DMX channel table.

DEVICE TYPE:

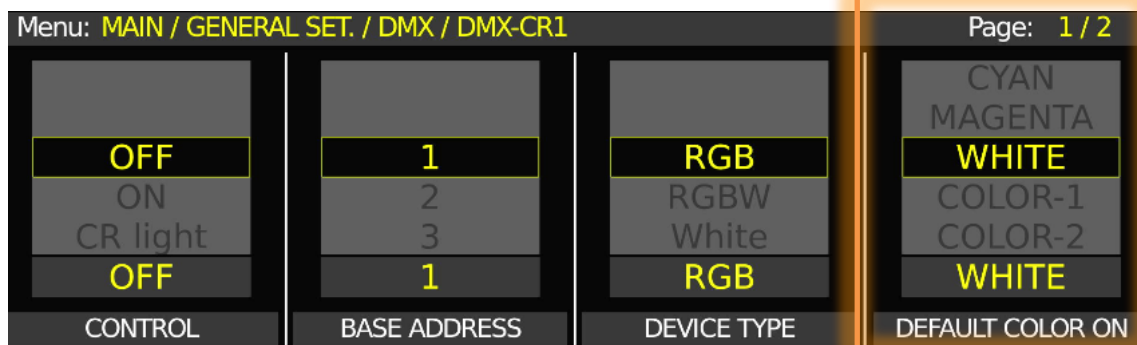


DEVICE TYPE selects the DMX profile that corresponds to the connected lighting fixture.

In this system, the available **DEVICE TYPE** options are:

- **RGB:** The fixture uses three consecutive channels for: **Red - Green – Blue**. Colours are created by combining the intensity of these three components.
- **RGBW:** The fixture uses four channels: **Red – Green – Blue – White**. The additional White channel can improve neutral-white reproduction and overall brightness.
- **WHITE:** The fixture is controlled through a single white-light channel. In this mode: The output is a single intensity value - Colour selection is not applied - **DEFAULT COLOR** settings are ignored or converted to intensity behaviour depending on firmware implementation.

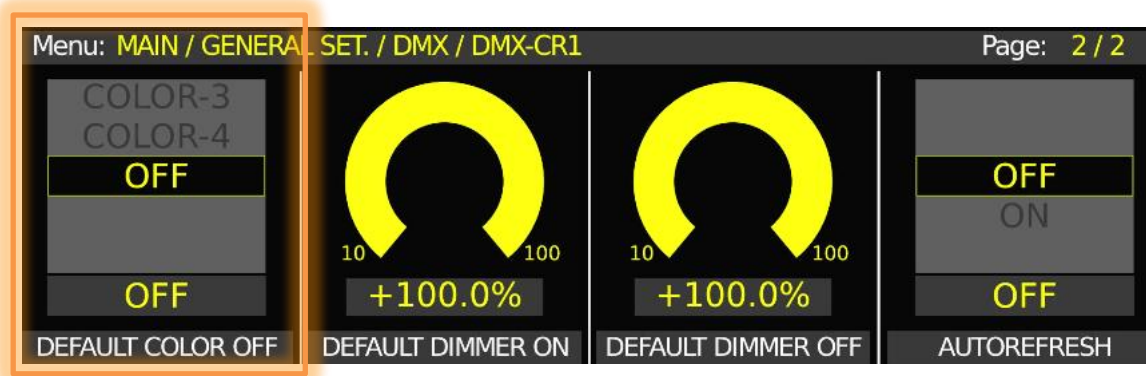
DEFAULT COLOR ON:



DEFAULT COLOR ON selects the colour transmitted when the associated Control Room lighting condition is **active**. **Selectable setting:**

- One colour from the available system colour palette
- A user-defined custom colour, when supported
- **OFF**

The selected colour is converted into the DMX values required by the **DEVICE TYPE**. For RGB or RGBW devices, the colour is translated into the corresponding channel values. For WHITE devices, the colour selection is typically converted into an equivalent intensity level.

DEFAULT COLOR OFF:

DEFAULT COLOR OFF selects the colour transmitted when the associated Control Room event is **inactive**.

Selectable setting:

- One colour from the available system colour palette
- A custom colour, when supported
- **OFF**

This parameter can be used to create a recognisable standby state. **DEFAULT COLOR OFF** does not necessarily switch the lamp off.

DEFAULT DIMMER ON:

DEFAULT DIMMER ON defines the brightness applied when the Control Room lighting condition is active.

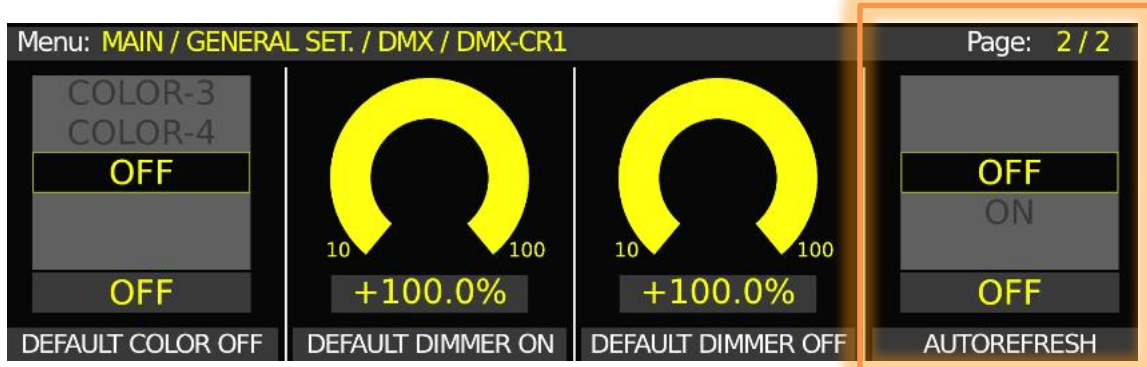
Range: 10.0% to 100.0%. Adjustment step: 1.0%. A high ON value is typically used for an On Air warning light, ensuring that the active indication is clearly visible. **Important:** maximum DMX value does not override the fixture's physical output limits, internal calibration or local brightness settings.

DEFAULT DIMMER OFF:



DEFAULT DIMMER OFF defines the brightness applied when the Control Room lighting condition becomes inactive. **Range: 10.0% to 100.0%. Adjustment step: 1.0%.** DEFAULT COLOR OFF and DEFAULT DIMMER OFF work together. If the dimmer is set to zero, the selected OFF colour is retained in the configuration but is not visibly displayed.

AUTOREFRESH:



AUTOREFRESH determines whether the OXYGEN 3000 Plus periodically retransmits the current DMX state. The selectable options are:

- **ON:** When AUTOREFRESH is set to **ON**, the console continuously or periodically resends the configured colour and dimmer values. This is generally the recommended setting for dedicated On Air and warning lights.
- **OFF:** When AUTOREFRESH is set to **OFF**, the DMX values are **normally transmitted only when:** The Control Room state changes - A relevant microphone event occurs - A configuration parameter is modified - The DMX position is enabled or activated. This reduces repeated control traffic, but a restarted fixture or DMX interface may not recover the correct state until the next lighting event occurs. **Important:** AUTOREFRESH does not create an animation or lighting effect. It only retransmits the current static DMX state.

DMX-ST1 / DMX-ST2:

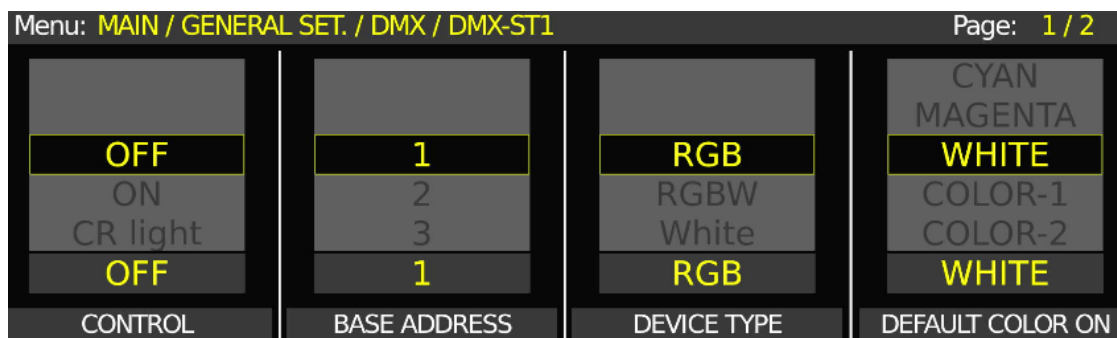
The **DMX-ST1** and **DMX-ST2** menus configure the two logical DMX lighting devices associated with the Studio.

- **DMX-ST1:** first Studio DMX device
- **DMX-ST2:** second Studio DMX device



Their primary purpose is to control compatible DMX lighting equipment according to the Studio's operating state. For example, the OXYGEN 3000 Plus can automatically change the colour or brightness of a lamp when a Studio microphone is activated. DMX-ST1 and DMX-ST2 are configured independently.

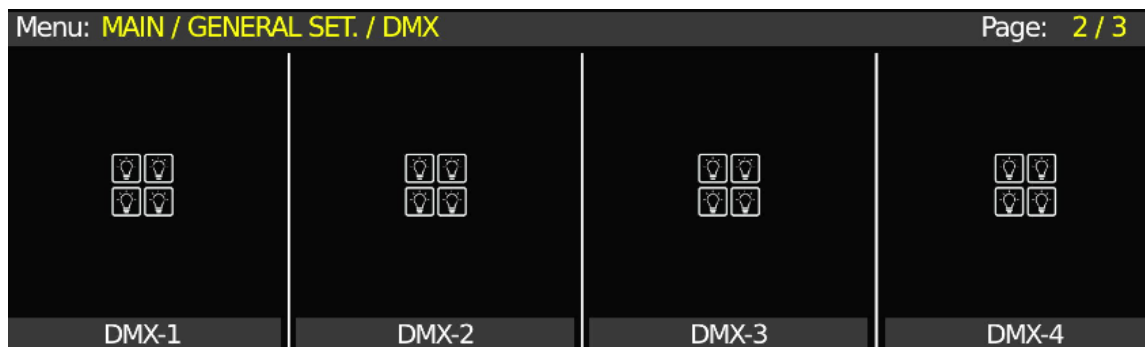
These are the editable options in this section:



ATTENTION: The options available in the **DMX-ST1/2** section are described in detail in the previous **DMX-CR1/2** subsection of the **DMX** section.

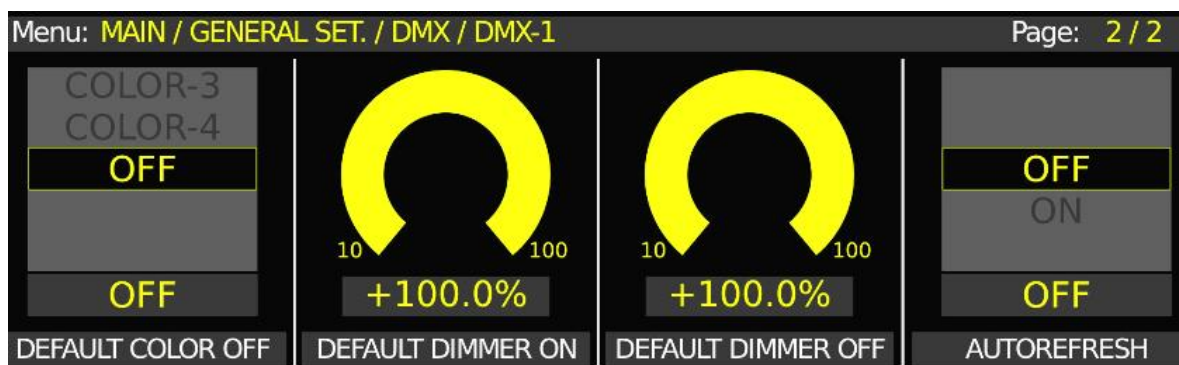
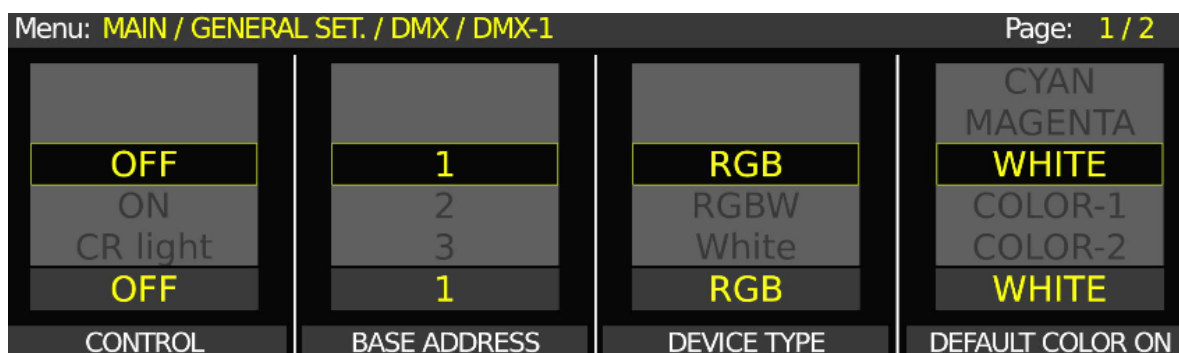
DMX 1–8:

DMX 1 through DMX 8 are eight independent, general-purpose DMX lighting positions provided by the OXYGEN 3000 Plus. Unlike **DMX-CR1/2** and **DMX-ST1/2**, these positions are not dedicated exclusively to the Control Room or Studio lighting logic. They can be assigned to individual input sources or used for other customised lighting functions. Each position is configured independently and can control a different DMX fixture, lighting zone or attribute.



Important: DMX 1–8 are logical control positions. They do not necessarily represent eight physical DMX connectors, devices or universes. The main purpose of DMX 1–8 is to associate external lighting equipment with specific console sources or operating events.

These are the editable options in this section:



ATTENTION: The options available in the **DMX 1-8** section are described in detail in the previous **DMX-CR1/2** subsection of the **DMX** section.

7. SMART KEYS

The **SMART KEYS** section configures the eight programmable function buttons identified as **K1 through K8** on the OXYGEN 3000 Plus. Smart Keys allow the console to send outgoing IP commands to compatible external devices or applications. **The SMART KEYS menu contains:**

- **BUTTON COLOR**
- **K1-8**

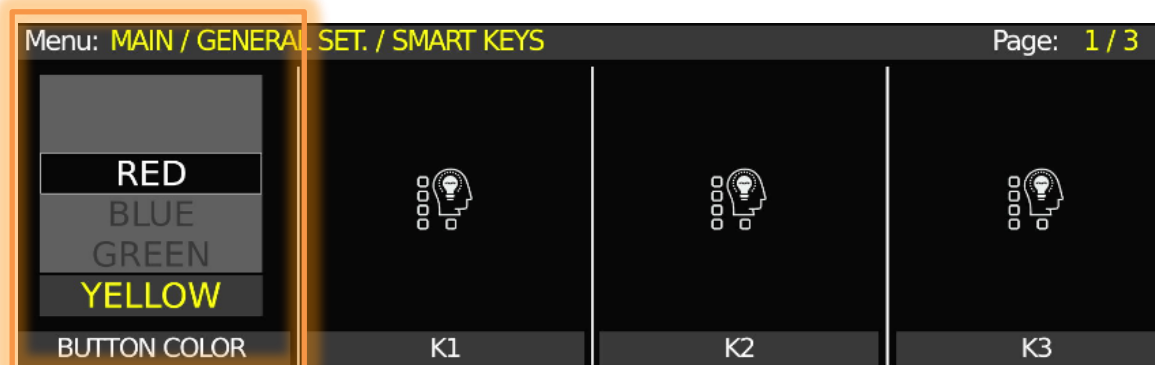


Important: the command, target device, communication protocol and operating mode of each Smart Key must be configured through **OXYGEN Remoter**. The console's SMART KEYS menu is primarily used to configure the buttons' colours.

BUTTON COLOR:

BUTTON COLOR selects the illumination colour assigned to the Smart Keys. The selectable options include all standard system colours, user-defined custom colours, and the OFF state. The available selections are:

- **RED**
- **BLUE**
- **GREEN**
- **YELLOW**
- **CYAN**
- **CUSTOM COLORS** (user-defined colours configured in the system palette)
- **OFF**

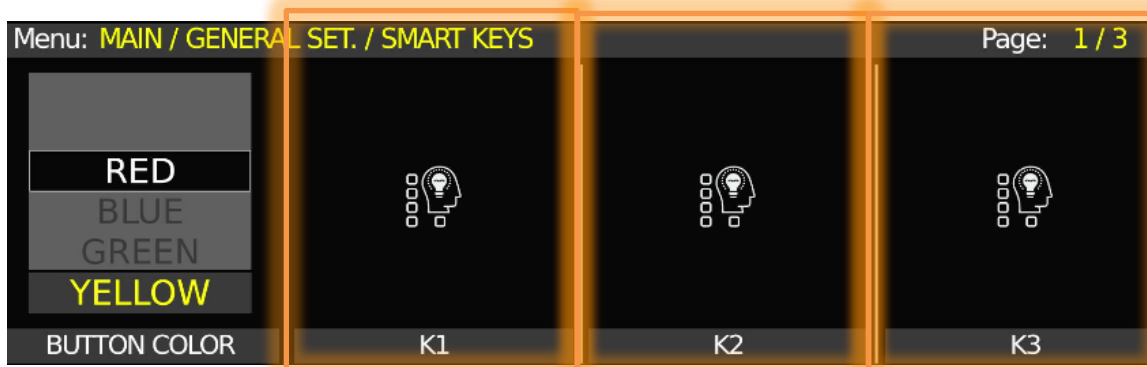


Different colours can be used to identify different categories of commands or operational states. The **OFF** option disables illumination of the Smart Key, leaving the button visually inactive regardless of its

assigned function. These are only suggested applications. The operator can define a different colour convention according to the station's operational standards. **Important:** changing the button colour does not modify the command assigned to the Smart Key and does not enable an unconfigured key.

K1–K8:

K1 through K8 identify the eight independently programmable Smart Key buttons.



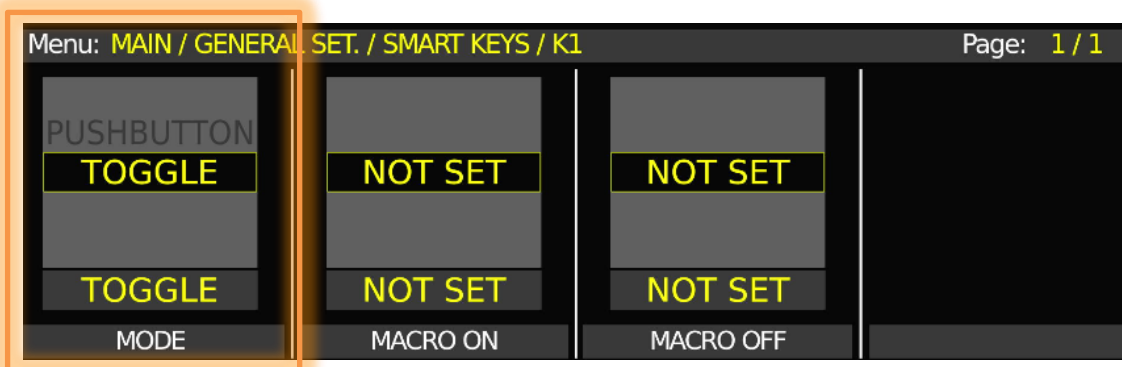
Each button provides three main configuration parameters:

- **MODE**
- **MACRO ON**
- **MACRO OFF**

Each Smart Key operates independently, allowing each button to control a different function, device or application.

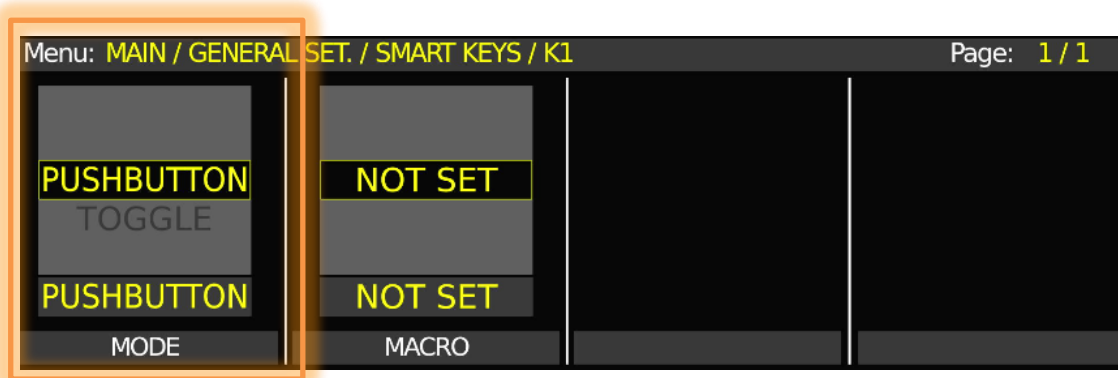
MODE:

MODE defines the operating behaviour of the Smart Key.



The available options are:

- **TOGGLE:** In **TOGGLE** mode, the Smart Key alternates between two states: **First press:** executes **MACRO ON** / **Second press:** executes **MACRO OFF**. Both macros can be independently configured. This mode is used for functions that require a persistent state change, such as: **START / STOP**. The button visually reflects the active state until the opposite command is executed.
- **PUSH BUTTON:** In **PUSH BUTTON** mode, the Smart Key behaves as a momentary control: **MACRO ON** is executed only while the button is pressed / Releasing the **button immediately stops the action** / **MACRO OFF** is not used. This mode is suitable for real-time or temporary actions



MACRO ON:

MACRO ON defines the command executed when the Smart Key is activated.

Depending on the selected **MODE**:

- In **TOGGLE mode**, **MACRO ON** is executed on the first press
- In **PUSH BUTTON mode**, **MACRO ON** is executed while the button is held

ATTENTION: The macros used by **SMART KEYS** must be created and configured through the **OXYGEN Remoter application or WEB PAGE**. The configuration path is: **OXYGEN REMOTER → SMART KEYS → SET**

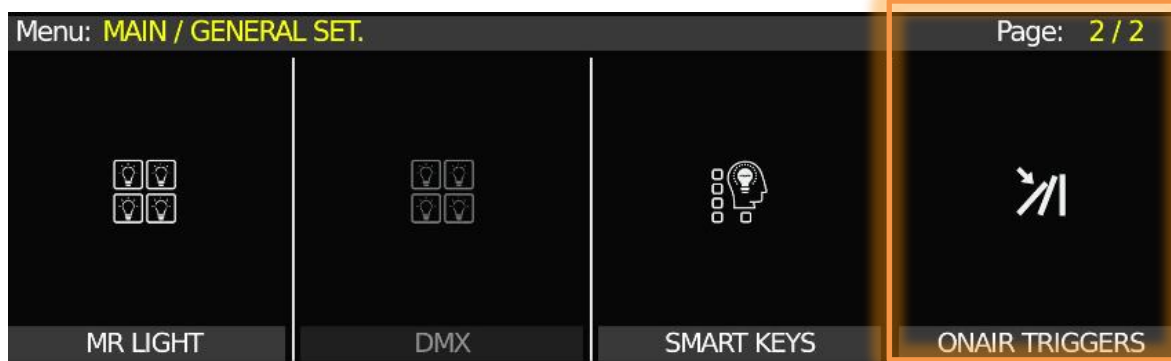
MACRO OFF:



MACRO OFF defines the command executed when the Smart Key is deactivated. This parameter is available only in **TOGGLE mode**. **MACRO OFF** typically contains the complementary action to **MACRO ON**. In **PUSH BUTTON mode**, **MACRO OFF** is not used, as the function is inherently momentary.

8. ONAIR TRIGGERS

The **ONAIR TRIGGERS** section associates an audio source with one or two IP-control macros. These macros are sent automatically when the selected source is activated or deactivated through its channel controls. Unlike **SMART KEYS**, which require the operator to press one of the K1–K8 buttons manually, ONAIR TRIGGERS operate automatically according to the status of the selected source.

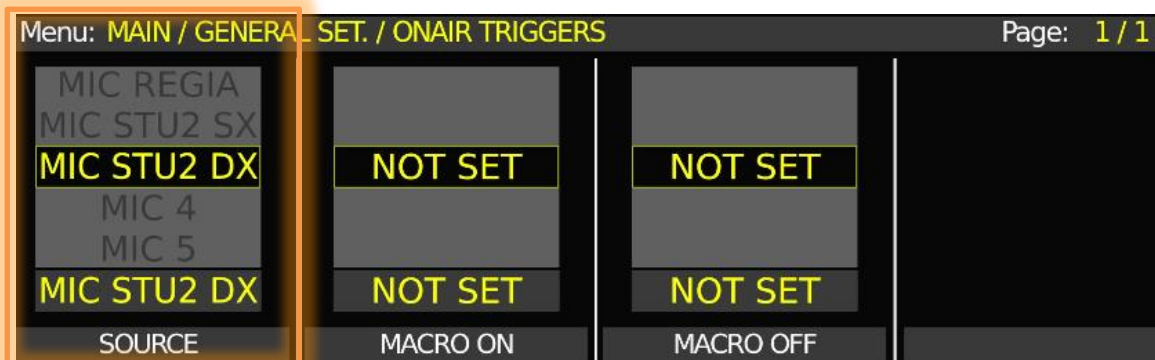


The ONAIR TRIGGERS menu contains:

- SOURCE
- MACRO ON
- MACRO OFF

INPUT SETTINGS (ONAIR TRIGGERS)

SOURCE:



SOURCE selects the audio input whose operational status must generate the trigger commands.

The selectable list is generated according to the **installed hardware, Input Mode settings, enabled options and firmware version**. It may include:

- MIC inputs
- MIC AUTOMIX
- MONO inputs
- STEREO inputs
- AES/EBU
- USB audio sources

- **BT sources**
- **PHONE and TELCO sources**
- **VoIP and Talk Show sources**
- **STREAMING sources**
- **DANTE ST, DANTE MONO and DANTE TELCO sources**
- **TONE GEN**

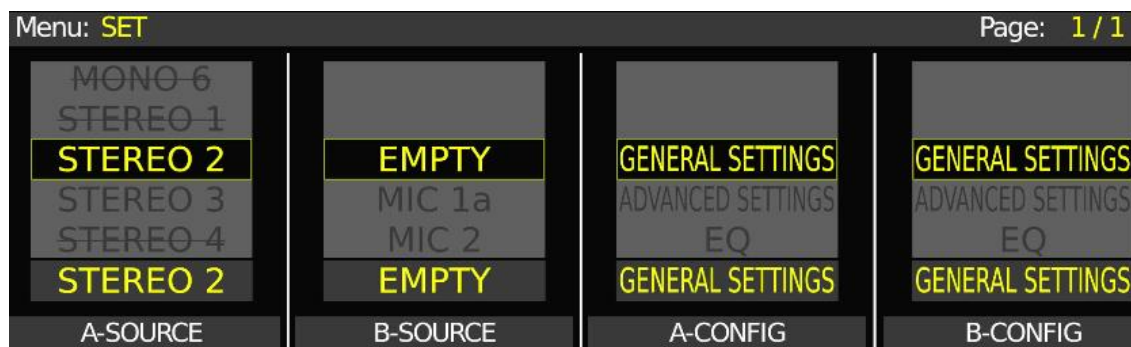
The trigger is associated with the **audio source**, rather than permanently with a specific physical fader.

If the selected source is assigned as the active **A-SOURCE** or **B-SOURCE** of a channel, the trigger follows the channel on which that source is currently active.

For example:

- **A-SOURCE:** MIC 1
- **B-SOURCE:** STEREO 1

When MIC 1 is selected as the active source, only the triggers assigned to **MIC 1** are used. If the channel is switched to STEREO 1, the console uses the triggers assigned to **STEREO 1**.



Important: if the same source is assigned to multiple channels, each eligible channel operation may generate the corresponding trigger. The configuration should therefore be checked carefully to prevent unintended repeated commands.

ATTENTION: The GAIN encoder associated with the channel can normally be pressed to switch between CH-A and CH-B. Rotating the encoder adjusts the gain of the currently selected source.

MACRO ON:

MACRO ON selects the macro sent when the chosen SOURCE enters its active state.

Depending on the source's **ON MODULE** configuration, the activation event may be generated when:

- The channel fader is raised beyond its opening threshold
- The associated **ON/START** button is activated
- The source enters the ON state through the configured fader-and-button logic

Selectable settings:

- **Any available macro previously created in OXYGEN Remoter**
- **No macro assigned**, when the corresponding empty or unassigned option is available



ATTENTION: The macros used by **ONAIR TRIGGER** must be created and configured through the **OXYGEN Remoter application or WEB PAGE**. The configuration path is:
OXYGEN REMOTER → SMART KEYS → SET

MACRO OFF:



MACRO OFF selects the macro sent when the chosen **SOURCE** returns to its inactive state. Depending on the source's **ON MODULE** configuration, the deactivation event may occur when:

- The channel fader is lowered below its closing threshold
- The associated **ON/START** button is switched off
- The source leaves the active state through the configured channel-control logic

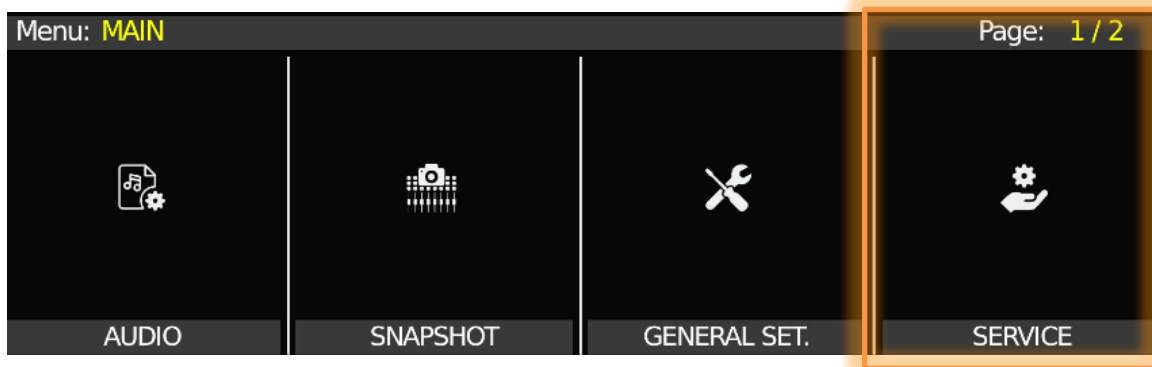
Selectable settings:

- **Any available macro previously created in OXYGEN Remoter**
- **No macro assigned**, when an **OFF** command is not required

MACRO OFF normally contains the complementary command to **MACRO ON**. **MACRO OFF** can be left unassigned when the external operation requires only a single activation command.

SERVICE

The **SERVICE** section, located inside the **MAIN** menu, contains the system-maintenance, configuration-management, security and diagnostic functions of the OXYGEN 3000 Plus. It can be accessed through: **MAIN → SERVICE**



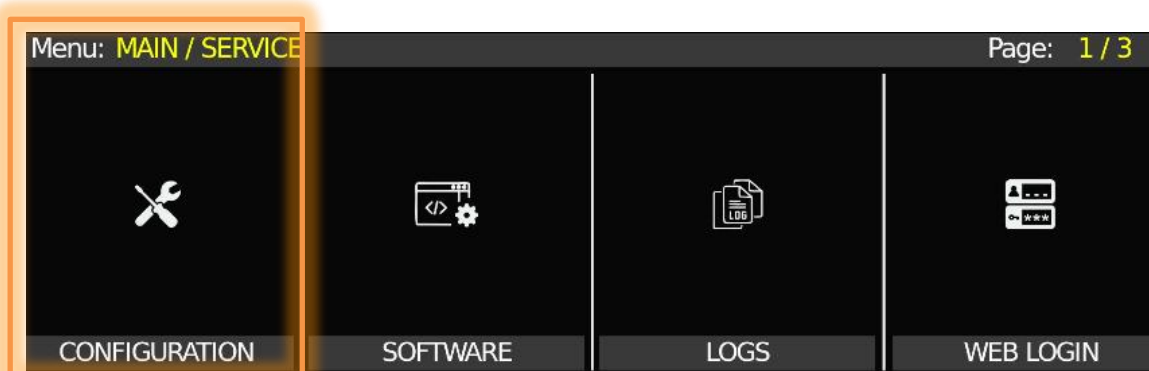
Unlike the **AUDIO** and **GENERAL SETTINGS** sections, **SERVICE** is not primarily used to configure normal input, output or routing operations. It is intended for activities such as:

- Saving and restoring the mixer configuration
- Updating the console firmware
- Checking software-version information
- Viewing and exporting system logs
- Managing web-access credentials

ATTENTION: some **SERVICE** operations can delete settings, restart parts of the console, interrupt remote access or temporarily make the mixer unavailable. Maintenance procedures should therefore be performed by qualified personnel and, when indicated, while the console is **not On Air**.

1. CONFIGURATION

The **CONFIGURATION** menu, located under **SERVICE**, manages the complete backup, recovery and reset functions of the OXYGEN 3000 Plus. One of the characteristics of OXYGEN is that **all console settings can be saved as a complete configuration**.

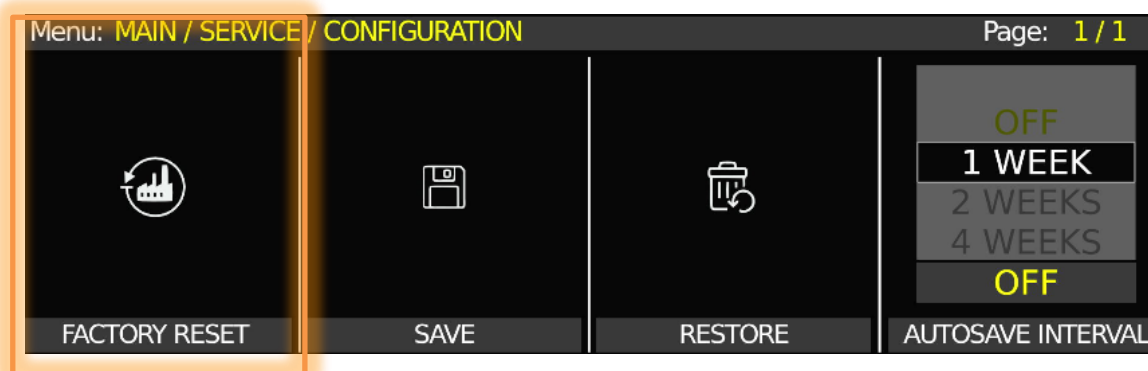


The saved state can later be recalled when required or exported to an external USB memory stick. A configuration exported to USB can also be transferred from one OXYGEN mixer to another, provided that the destination console has a **compatible hardware configuration and firmware version**.

The CONFIGURATION menu includes:

- **FACTORY RESET**
- **SAVE**
- **RESTORE**
- **AUTOSAVE INTERVAL**

FACTORY RESET:

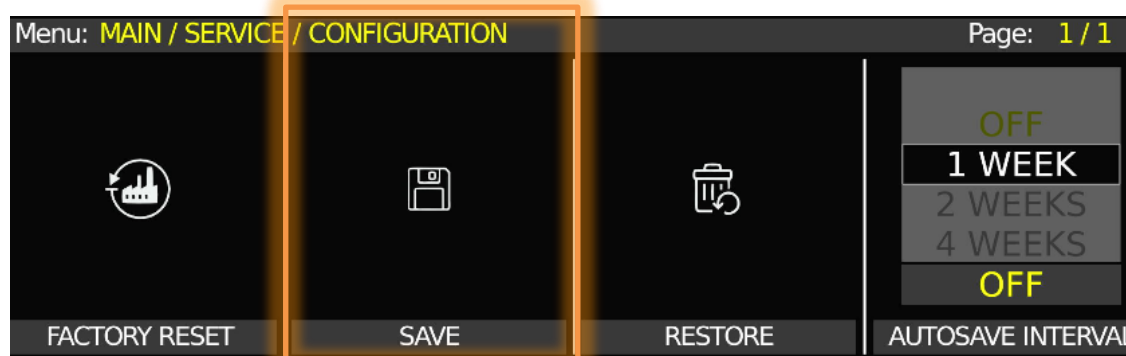


FACTORY RESET restores the mixer to its original manufacturer configuration by deleting the settings and information previously stored on the console. When FACTORY RESET is selected, the console displays a confirmation request to prevent the configuration from being deleted accidentally. The available actions are:

- **CONFIRM / YES:** executes the Factory Reset and deletes the stored settings.
- **CANCEL / NO:** stops the procedure without modifying the console.

A Factory Reset does not restore a user-created backup. It loads the default configuration provided by the manufacturer.

Important: FACTORY RESET is a destructive operation. Before confirming it, save the current configuration to the internal SD memory and export an additional copy to a USB memory stick whenever the existing settings may be needed again.

SAVE:

SAVE stores the current system state of the OXYGEN 3000 Plus. Saving the system state creates a complete configuration backup representing the console at that particular point in time.

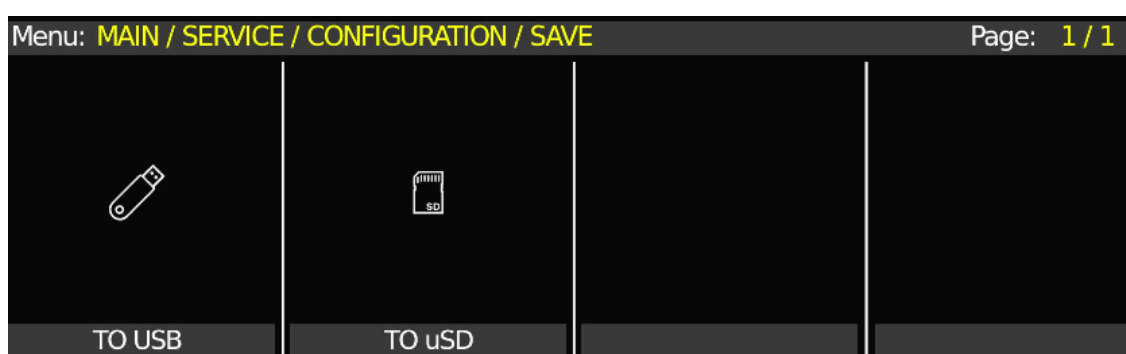
When **SAVE** is selected from the console, the available destinations are:

- **INTERNAL SD:** saves the configuration to the mixer's internal SD memory.
- **USB / EXTERNAL MEMORY:** exports the configuration to a USB memory stick connected to the console.

The **internal SD memory** provides a backup that remains directly available on the mixer. The **USB option** creates a portable copy that can be stored separately or imported into another compatible OXYGEN mixer. When configuration management is performed through the console's Web interface, an additional destination may be available:

- **LOCAL PC:** downloads the configuration file to the computer currently connected to the Web interface.

Important: do not remove the USB memory stick or switch off the mixer while the configuration is being saved. Wait for the operation-completed indication before disconnecting the device.

**Configuration transfer between mixers:**

A configuration exported to a USB memory stick can be used to reproduce the same operating setup on another compatible OXYGEN mixer. A typical transfer procedure is:

1. Insert a USB memory stick into the source mixer.
2. Open **SERVICE → CONFIGURATION → SAVE**.
3. Select **USB / EXTERNAL MEMORY**.
4. Wait for the saving procedure to finish.
5. Remove the USB memory stick safely.
6. Connect it to the destination mixer.

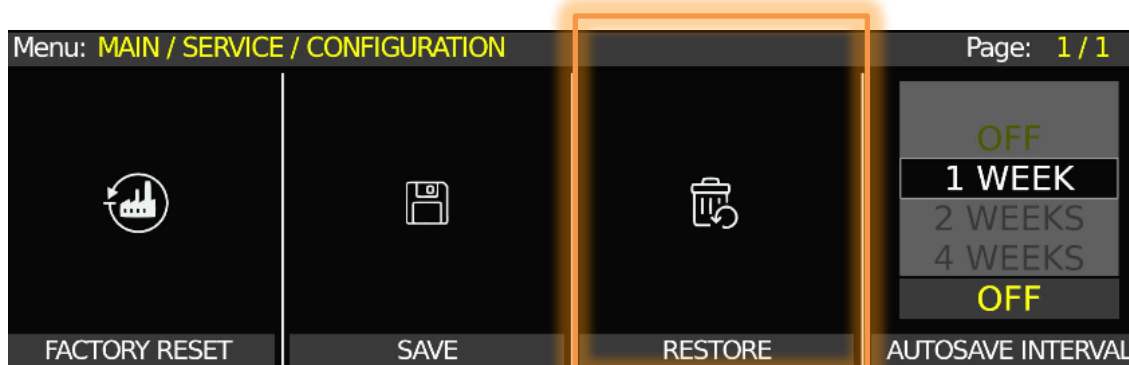
7. Open **SERVICE → CONFIGURATION → RESTORE**.
8. Select the required configuration file and confirm the restoration.

Compatibility note: before transferring a configuration, verify that the source and destination mixers have compatible numbers of faders, Audio I/O Boards, optional interfaces and firmware versions. Parameters associated with unavailable hardware may not be restored correctly.

RESTORE:

RESTORE recalls a previously saved system state and applies it to the console.

When entering this section, the mixer displays the available saved configurations, normally organised according to their **saving date and time**. This information helps identify the required recovery point.



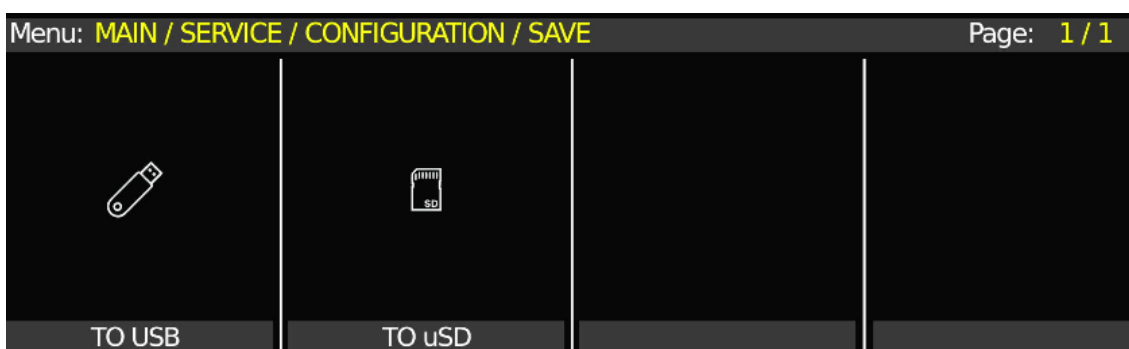
Depending on where the backup was stored, the selectable restore sources are:

- **INTERNAL SD:** displays configurations saved in the mixer's internal SD memory.
- **USB / EXTERNAL MEMORY:** displays compatible configuration files stored on the connected USB memory stick.

When using the Web interface, configurations may also be restored from:

- **LOCAL PC:** uploads a previously downloaded configuration file from the connected computer.

After selecting the storage source, the operator selects the required configuration from the available list and confirms the restore operation.



When a system state is saved, the console stores a representation of its settings and configuration data at that particular moment. **RESTORE** returns those parameters to their saved values.

Important: restoring a configuration replaces the current settings. If the present configuration may be required later, save it before starting the restore procedure.

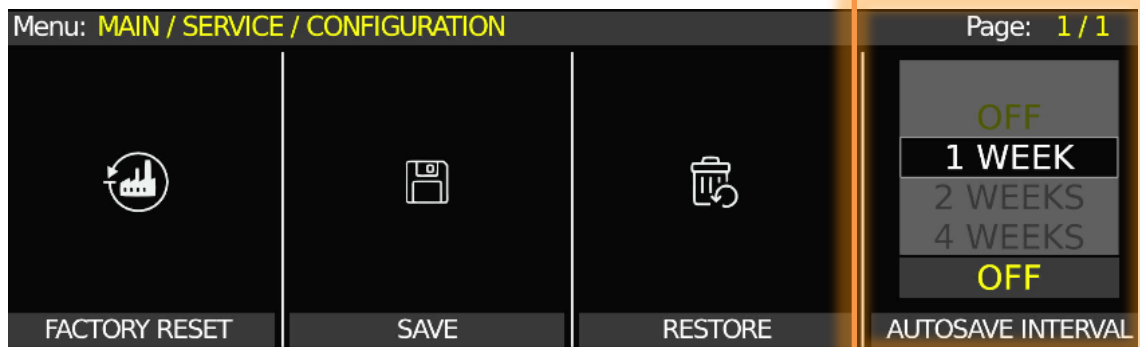
AUTOSAVE INTERVAL:

AUTOSAVE INTERVAL determines how frequently the console automatically records its current configuration state.

Automatic saving reduces the risk of losing recent changes if the mixer is unexpectedly switched off, restarted or affected by a system interruption.

The selectable settings normally include:

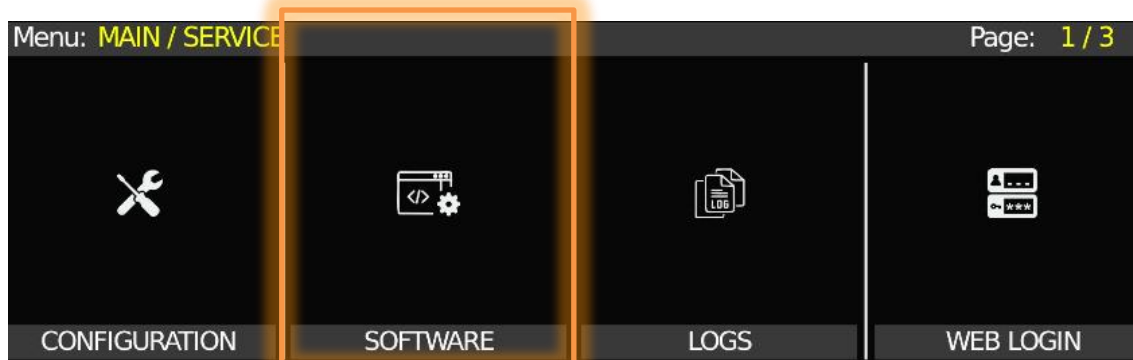
- **OFF / DISABLED:** disables periodic automatic configuration saving.
- **Available time intervals:** automatically saves the configuration at the selected frequency.



Important: Automatic saving is intended primarily for **local recovery**. It does not automatically export the configuration to a USB memory stick. Portable backups must be created manually through **SAVE → USB / EXTERNAL MEMORY**.

2. SOFTWARE

The **SOFTWARE** menu, located under **SERVICE**, manages the firmware of the OXYGEN 3000 Plus and displays information about the software currently installed on the console.



The section includes:

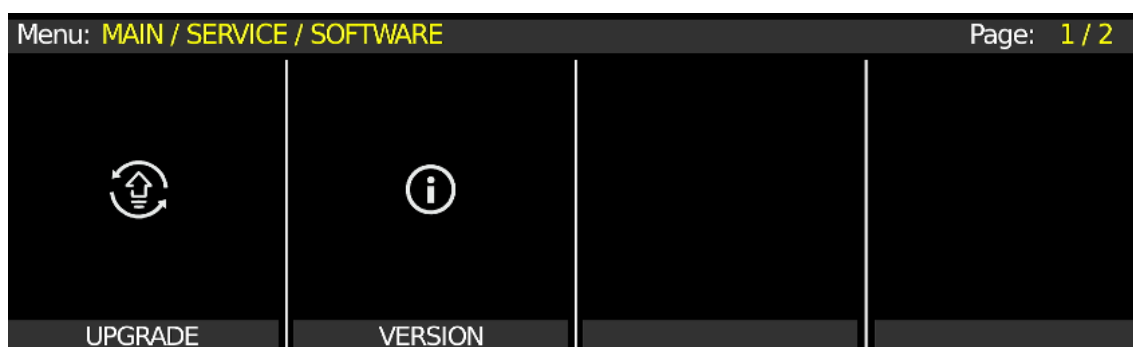
- **UPGRADE**
- **VERSION**
- **WATCHDOG**

The firmware controls the console's operating functions, menu structure, audio-management logic, hardware communication and compatibility with optional features.

ATTENTION: a firmware update must never be performed while the console is On Air. The mixer remains unavailable during part of the procedure and may restart automatically.

UPGRADE:

The **UPGRADE** menu, located under **SERVICE → SOFTWARE**, manages the download, preparation and installation of firmware updates for the OXYGEN 3000 Plus.



The console can obtain an update in two ways:

- **Online update:** the firmware package is downloaded directly from the Axel Technology server.
- **USB update:** the firmware package is read from an external USB memory stick.

Installed version	8.7.0.9 beta
Selected latest version	8.6.0.26

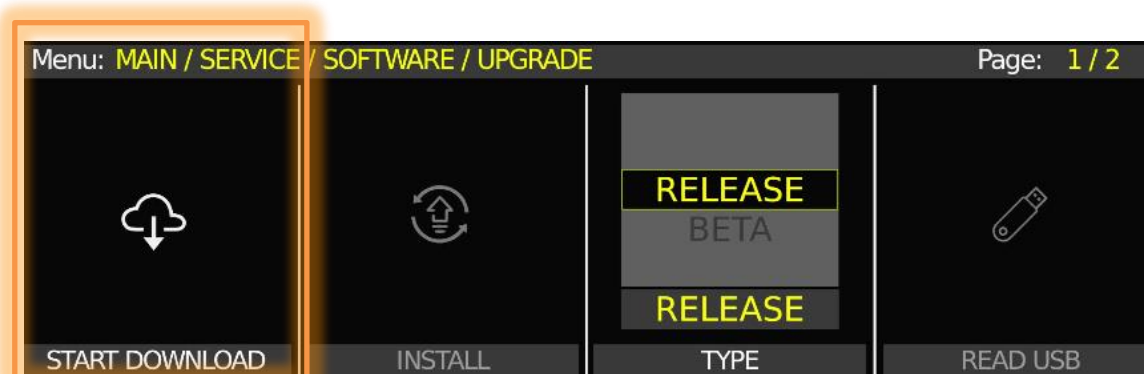
The UPGRADE menu includes:

- **START DOWNLOAD**
- **INSTALL**
- **TYPE**
- **READ USB**
- **FORCE SW UPDATE**
- **FORCE MICRO UPDATE**

The standard procedure normally uses **TYPE**, **START DOWNLOAD** and **INSTALL**. The USB and forced-update controls are intended mainly for offline maintenance, recovery procedures or operations requested by Axel Technology technical support.

ATTENTION: save the console configuration before starting any update. The mixer must not be updated while it is On Air, and its power supply must remain connected throughout the entire procedure.

START DOWNLOAD:



START DOWNLOAD downloads the selected firmware package directly from the Axel Technology update server. **Selectable operation:**

- **PRESS / START DOWNLOAD:** starts the online download.

Before selecting **START DOWNLOAD**, the operator must configure **TYPE** to choose the required update channel. The console must also have a working Internet connection. Network parameters can be configured through: **GENERAL SET. → COMMUNICATIONS → TCP-IP**

The connection may use:

- **DHCP:** automatically receives the IP address, subnet mask, gateway and DNS settings from the network.
- **Static IP:** uses network parameters entered manually.

During the download, the console normally displays a progress indicator or percentage. **START DOWNLOAD** retrieves and prepares the update package but does not immediately install it. When the preparation is complete, the console displays an upgrade-ready indication and makes **INSTALL** available.

Important: do not switch off the console, disconnect the network or start another maintenance operation while the package is being downloaded or prepared.

INSTALL:



INSTALL applies the firmware package that has already been downloaded from the Internet or successfully read from a USB memory stick. **Selectable operation:**

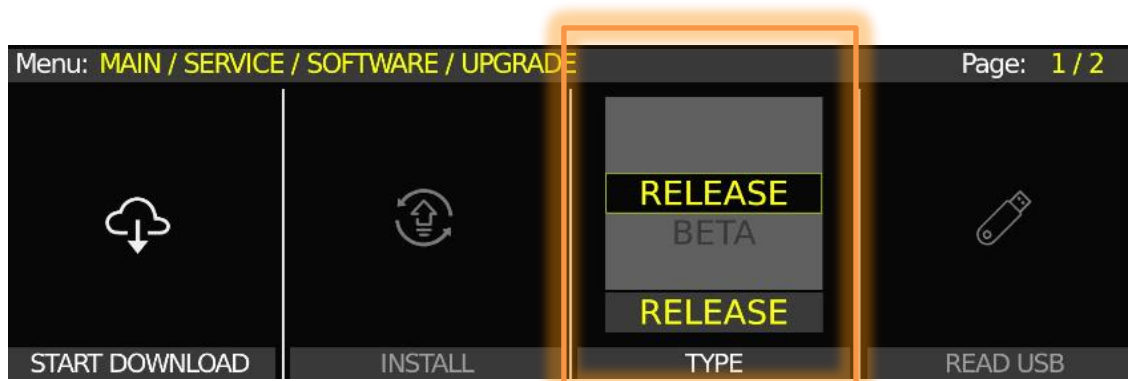
- **PRESS / INSTALL:** starts installation of the prepared firmware package.

INSTALL normally remains inactive until a valid update package has been prepared through either:

- **START DOWNLOAD**
- **READ USB**

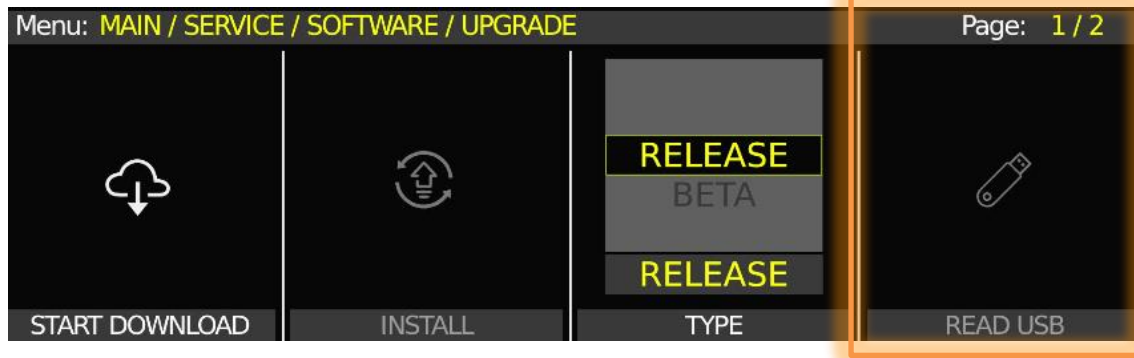
The standard documented update procedure takes approximately **six minutes**, although the actual duration depends on the firmware package and the console's hardware configuration. **Warning:** never disconnect the power supply or interrupt the console while INSTALL is running. An interrupted installation may leave the mixer unable to start correctly and may require a technical recovery procedure.

TYPE:



TYPE selects the online firmware channel used by START DOWNLOAD. The selectable options are:

- **RELEASE: RELEASE** selects the latest officially released firmware available for the OXYGEN 3000 Plus. Release firmware has completed the normal validation process and is intended for standard operational use. **RELEASE is the recommended selection for normal production and On Air systems.**
- **BETA: BETA:** selects a pre-release firmware package made available for testing, technical evaluation or specific maintenance procedures. A Beta version may contain new functions or corrections that have not yet completed the complete release-validation process. Changing TYPE does not update the mixer immediately. It only determines which package is retrieved when START DOWNLOAD is selected.

READ USB:

READ USB searches an external USB memory stick for a compatible firmware-update package and prepares it for installation. **Selectable operation:**

- **PRESS / READ USB:** scans the connected USB memory stick and reads the update package.

A typical USB update procedure is:

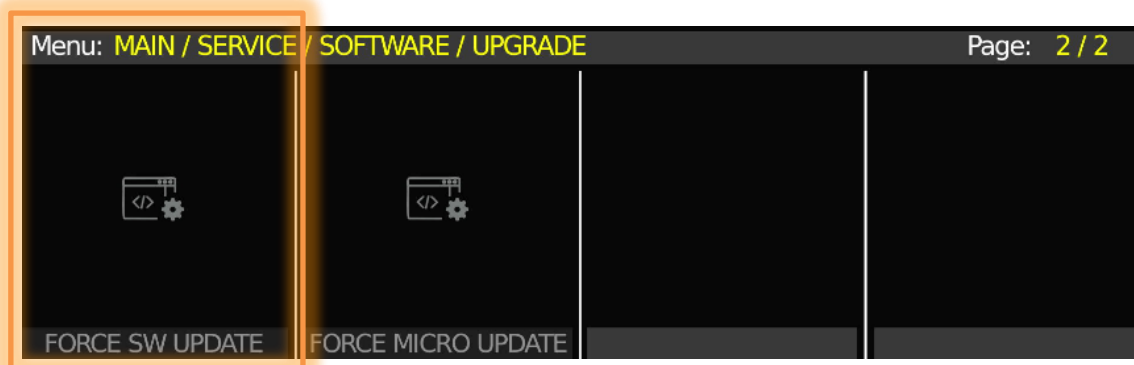
1. Obtain the correct update package from Axel Technology or an authorised service centre.
2. Copy the package to a compatible USB memory stick without changing its required filename or folder structure.
3. Insert the memory stick into one of the console's USB Type-A ports.
4. Open **SERVICE → SOFTWARE → UPGRADE**.
5. Select **READ USB**.
6. Wait while the console reads and validates the package.
7. Verify that the package has been recognised successfully.
8. Select **INSTALL** to start the update.

If no compatible package is found, **INSTALL** should remain unavailable and the console may display an error or file-not-found indication. **Important:** use only firmware packages specifically intended for the OXYGEN 3000 Plus. Do not rename, modify or extract the update files unless the instructions supplied with the package explicitly require it.

FORCE SW UPDATE:

FORCE SW UPDATE controls whether the console must reinstall the main system software even when the normal update logic would not consider the operation necessary. This option may be useful after an interrupted update or when instructed by technical support to reinstall a specific software version.

Warning: **FORCE SW UPDATE** bypasses part of the normal decision logic. It must not be used routinely and should remain **OFF** unless a forced reinstallation is specifically required.



FORCE MICRO UPDATE:

FORCE MICRO UPDATE controls the forced reprogramming of the firmware used by the console's supported internal microcontrollers. These microcontrollers manage communication and operating functions associated with the console's internal hardware modules and control surface. A forced microcontroller update may take longer than a normal software-only operation and may cause several internal modules or surface sections to restart.

Warning: FORCE MICRO UPDATE is an advanced service function. An interruption during microcontroller programming can prevent an internal module or the control surface from operating correctly. Use this option only with a stable power supply and preferably under instructions from Axel Technology technical support.



VERSION:

VERSION displays information about the software and firmware currently installed on the OXYGEN 3000 Plus. The principal displayed item is:

- **FIRMWARE**

Depending on the installed firmware and hardware configuration, the page may also show version identifiers for individual console components or software modules.

Oxygen 3000+ 2 Audio Dante 16Ch



Software and firmware version

Environment	8.7.0.9 beta	BRIDGE	2.3
GUI	2.10.0.1	MASTER MCU	5.3
Bridge status	Ready	DANTE1 DSP	5.1
FPGA	2.0		
AUDIO1 DSP	4.1		
AUDIO2 DSP	4.1	CH1 MCU	5.4
AUDIO1 MCU	4.1	CH2 MCU	5.4
AUDIO2 MCU	4.1	CH3 MCU	5.4
MATRIX DSP	5.1	CH4 MCU	5.4

Activated modules:	Talk show, DMX, VOIP		
MAC Address	88:A2:9E:B2:B2:98		
Serial Number	0000000000000000	CPU	67,7°C
Uptime	01:28:41		17,5%

Important: VERSION reports the software currently installed on the console, whereas the version shown inside UPGRADE may refer to the firmware package available for download. These two values may be different when a newer update is available.

WATCHDOG:

WATCHDOG is a system supervision function designed to monitor the correct execution of the console's internal software processes and ensure automatic recovery in case of abnormal behaviour.



When enabled, the WATCHDOG continuously checks the system status. If a software process becomes unresponsive or a critical error is detected, the system can automatically trigger a controlled recovery action, such as a restart of the affected module or of the entire console. The WATCHDOG function is intended to increase operational reliability, especially in continuous On Air environments.



ENABLE:



This option activates or deactivates the WATCHDOG supervision system.

- **ON:** activates continuous system monitoring.
- **OFF:** deactivates WATCHDOG supervision.

Important: disabling WATCHDOG is not recommended in production environments, as it removes automatic recovery mechanisms in case of system malfunction.

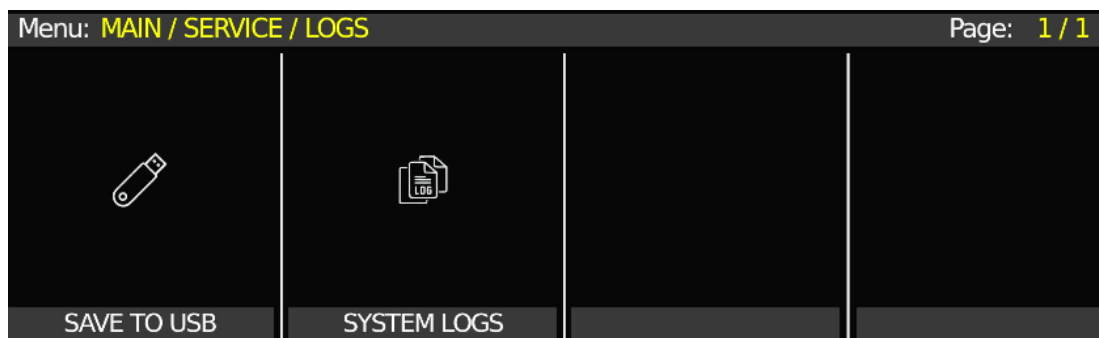
3. LOGS

The **LOGS** menu, located under **SERVICE**, provides access to the operating records generated automatically by the OXYGEN 3000 Plus.

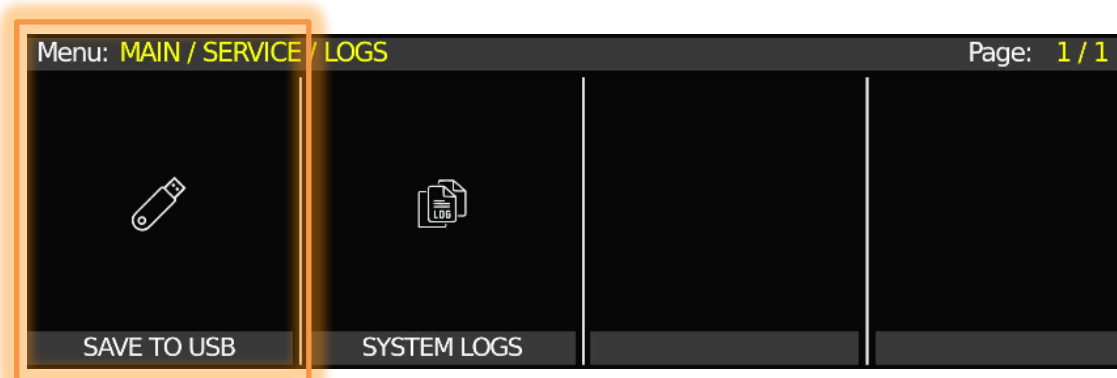


A log is a chronological record of events occurring during console operation. These records help technicians identify the conditions associated with an error, unexpected restart, communication problem or maintenance operation. The LOGS menu includes:

- **SAVE TO USB**
- **SYSTEM LOGS**



SAVE TO USB:



SAVE TO USB exports the console's available log data to an external USB memory stick. **SAVE TO USB** is an execution command rather than a permanent operating setting. **Selectable operation:**

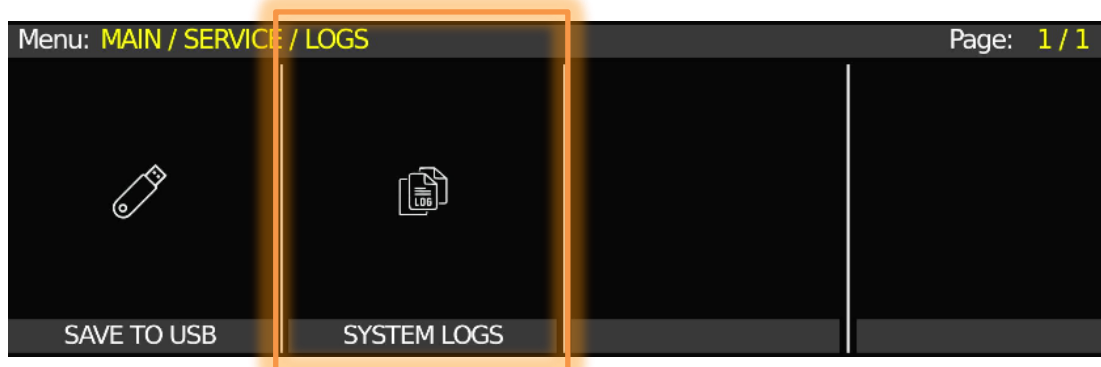
- **PRESS / SAVE TO USB:** starts exporting the available logs to the connected USB memory stick.

The console normally generates the required filename and file structure automatically. The exported data may be stored as a text file or another diagnostic format defined by the installed firmware. **Important:** do not remove the USB memory stick or switch off the console while the export is running. Interrupting the operation may produce an incomplete or unreadable log file.

SYSTEM LOGS:

SYSTEM LOGS opens the system-event history stored by the OXYGEN 3000 Plus.

This section is a **read-only diagnostic viewer**. It allows the operator or technician to examine the events recorded by the console without modifying its operating configuration.



Access through Remoter and Web interface

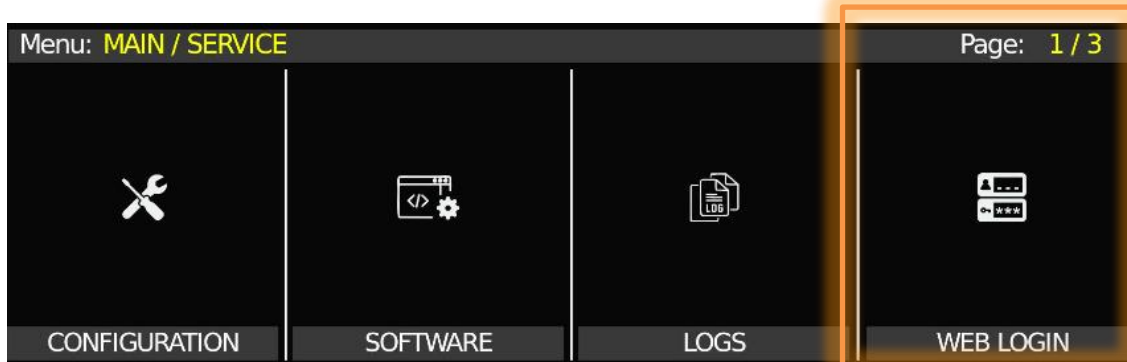
System logs may also be accessed through:

- **OXYGEN Remoter**
- **The console Web interface**

The Web interface can provide additional navigation functions, such as selecting a date from a calendar, reading the related records and downloading the log as a text file. These interfaces display information generated by the same console logging system; they do not replace the records stored by the mixer.

4. WEB LOGIN

The **WEB LOGIN** menu, located under **SERVICE**, configures the password used to access the OXYGEN 3000 Plus integrated Web interface.



The Web interface allows authorised users to connect to the console through a browser and access functions such as:

- Console configuration
- Audio and general settings
- Configuration backup and restore
- System-log consultation and export
- Firmware and service information
- Remote maintenance functions supported by the installed firmware

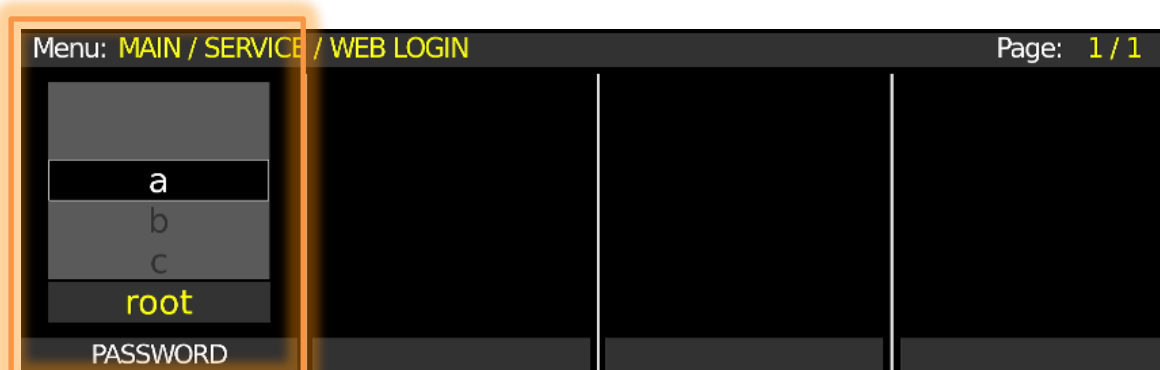
To open the Web interface, enter one of the console's configured IP addresses in the address bar of a computer connected to the same reachable network. The WEB LOGIN section principally contains:

- **PASSWORD**

It also provides the character-selection and deletion controls required to enter the new password.

Important: WEB LOGIN changes the credential used by the integrated Web interface. It does not modify the console's IP address, the four-digit surface ACCESS CODE or the credentials used by other applications unless explicitly indicated by the firmware.

PASSWORD:



PASSWORD sets the password required to log in to the console's Web interface. The factory-default Web-interface password is:

- **root**

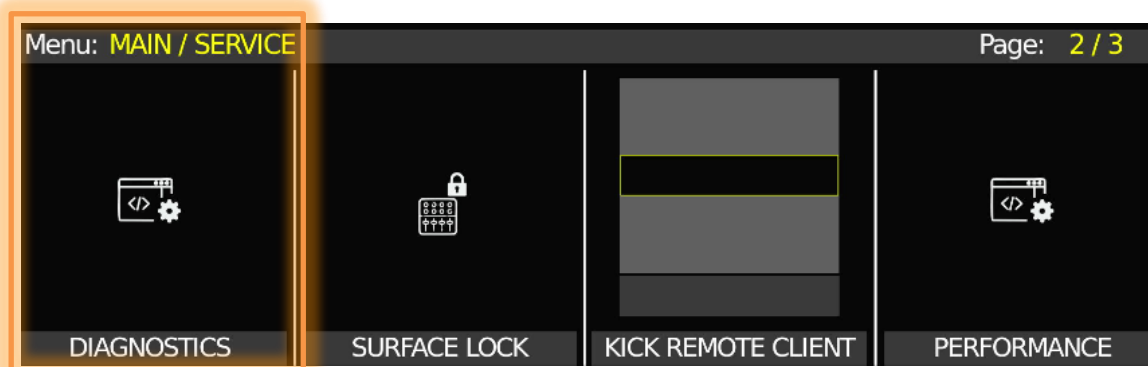
For security reasons, the default password should be replaced during the initial installation, especially when the console is connected to a shared network. **Selectable settings and operations include:**

- **Available characters:** selects the letters, numbers or symbols provided by the on-screen character list.
- **PRESS / CONFIRM CHARACTER:** inserts the selected character into the password.
- **<<:** deletes the last entered character.
- **CONFIRM / APPLY:** stores the completed password, when the corresponding confirmation is displayed by the firmware.

The available character set and maximum password length may vary according to the installed firmware version. **Important:** record the new password in a secure location before closing the menu. Losing the Web-interface password may require a recovery procedure or assistance from authorised technical support.

5. DIAGNOSTIC

The **DIAGNOSTIC** menu, located under **SERVICE**, provides maintenance and troubleshooting functions for the OXYGEN 3000 Plus.

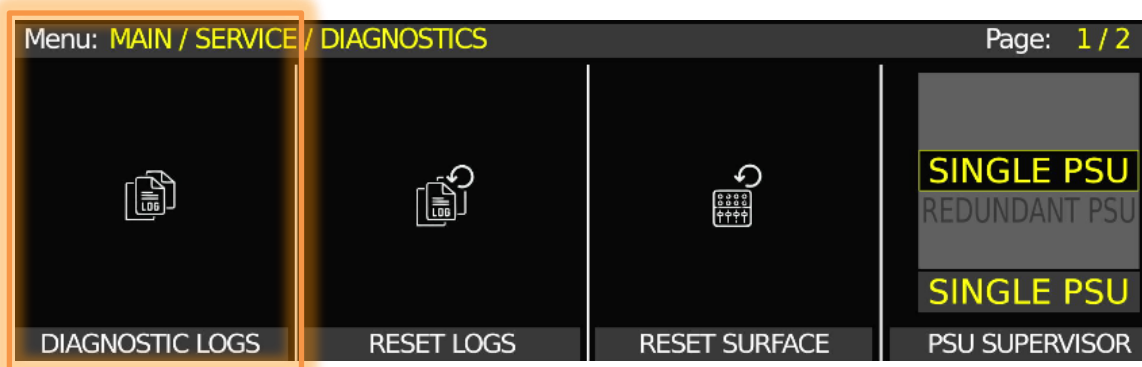


The DIAGNOSTIC section includes:

- **DIAGNOSTIC LOGS**
- **RESET LOGS**
- **RESET SURFACE**
- **PSU SUPERVISOR**
- **TEST CONFIGURATION**
- **START SURFACE TEST**

Important: diagnostic reset and surface-test functions should be used only by qualified personnel and while the console is **not On Air**. Some operations can temporarily disable the physical controls or remove information required for technical support.

DIAGNOSTIC LOGS:



DIAGNOSTIC LOGS opens the internal diagnostic-event viewer. The logs provide technical information about the operation of the mixer and may help identify intermittent or persistent problems. **DIAGNOSTIC LOGS** is principally a **read-only page**. Opening or examining an entry does not modify the console configuration. **Important:** diagnostic messages may contain technical codes intended for Axel Technology support. Record or export the relevant information before performing **RESET LOGS**.

Diagnostics logs

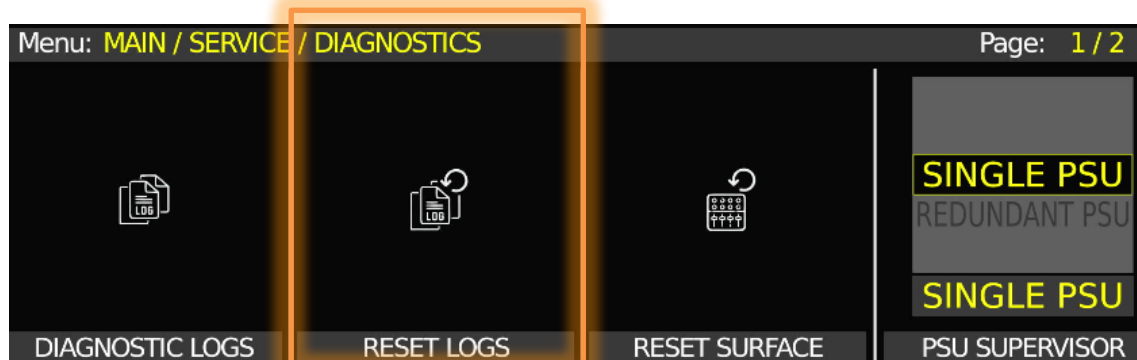
```
#Press any key to refresh:

Wireless network          C:\NET\3006...
last reset time           26/07/2025 14:48:57
system wireless mode      no wireless
stress vs internal state   no stress
stress vs external busness logic no stress
time from last boot start/reset 09/06/2025 00:00:00
include CPU load          12 (100%)
instant system memory      40
include Business logic Test packet loss 0 (0)
include Business logic Test packet Data Command Error 0 (0)
include Business logic Test packet latency 0 (0) -> 0.000ms
current average business logic test packet latency 2.4 (100%)
current communication link time 0 (0)
channel load buffer overflow 0 (0)
master input buffer overflow 0 (0)
ISP input buffer overflow 0 (0)
shared default buffer overflow 0 (0)
master default buffer overflow 0 (0)
last time via CPU load    10 (100%)
last time average CPU load 7 (70%)
last time System memory 41 (96%)
wireless business logic test packet loss 0 (0)
wireless business logic test packet data command error 0 (0)
wireless business logic test packet latency ssa 0.0 (100%)
wireless business logic test packet latency average 0.0 (100%)
test time communication link time 0.0 (10%)
number of SSID search without success 0 (0)
number of Device Failure Notifications 0 (0)

Device                               Vendor-FW
device "not found or bad" error      0 (0)
incorrect or corrupted packet on bus from device error 0 (0)
internal param error on received packet from device 0 (0)
ssd internal latency on received packets from device 0.0 (100%)
storage internal latency on received packet from device 0 (0)
internal communication data loss     0 (0)
internal device supply access anomaly 0 (0)
internal device surface frown anomaly 0 (0)
internal device general anomaly      0 (0)
internal device peripheral fw anomaly 0 (0)
internal device mem vs wll anomaly    0 (0)
win value for DV supply              4.9 (1.90)
win value for 3.3v Supply            2.1 (10.40)
hardware failure memory              0 (0)

Device                               AuditID [log]
device "not found or bad" error      0 (0)
incorrect or corrupted packet on bus from device error 0 (0)
internal param error on received packet from device 0 (0)
internal communication data loss     0 (0)
internal device general anomaly      0 (0)
internal device peripheral fw anomaly 0 (0)
hardware failure memory              0 (0)
```

RESET LOGS:



RESET LOGS deletes the diagnostic events currently stored by the console. This operation may be useful:

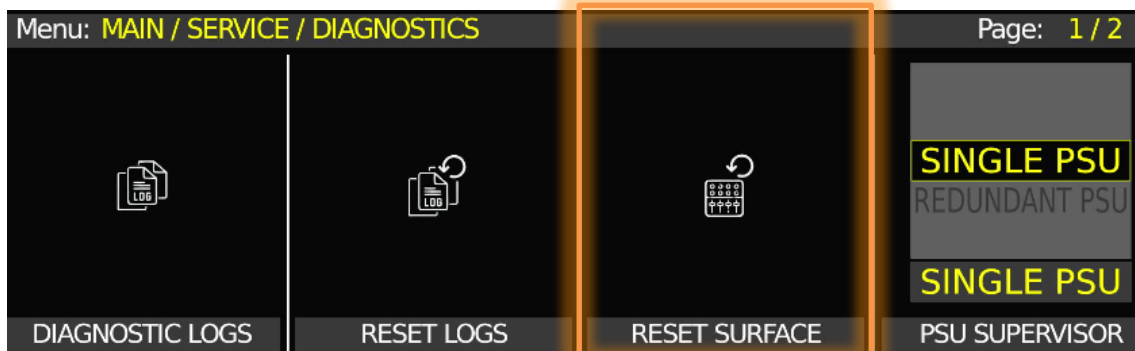
- After a completed maintenance intervention
- After diagnostic information has been exported
- Before reproducing a problem to obtain a clean event history
- When instructed by Axel Technology technical support
- When obsolete diagnostic records are no longer required

After RESET LOGS has been executed, the mixer begins recording new diagnostic events from an empty history. RESET LOGS **does not**:

- Restore factory settings
- Delete audio-source configurations
- Change routing or processing parameters
- Repair the condition that generated an error
- Reset the physical control surface

Warning: deleted diagnostic records may no longer be available for troubleshooting. Save or export the relevant logs before confirming the reset.

RESET SURFACE:

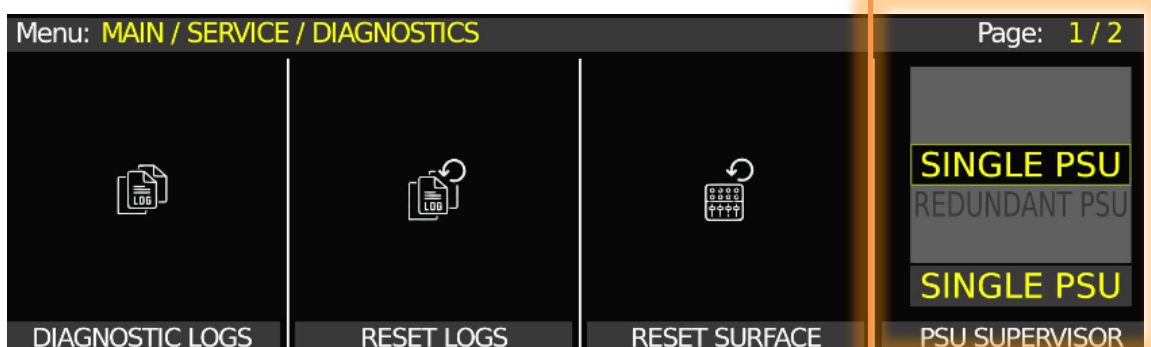


RESET SURFACE restarts or reinitialises the physical control-surface subsystem of the OXYGEN 3000 Plus. The control surface includes components such as:

- Faders
- Buttons and Smart Keys
- Rotary encoders
- LEDs and illuminated controls

This command can be used when the surface is not responding correctly, while the console software remains accessible through the main menu or a remote-control interface. **RESET SURFACE** is not equivalent to a **FACTORY RESET**. It does not normally delete console settings, saved configurations or audio routing. **Warning:** perform **RESET SURFACE** only when the console is not On Air. Audio paths may remain active in some conditions, but the operator temporarily loses reliable access to the physical controls; continuity must therefore not be assumed during the procedure.

PSU SUPERVISOR:



PSU SUPERVISOR defines the operating mode of the console's power-supply system. The available modes are:

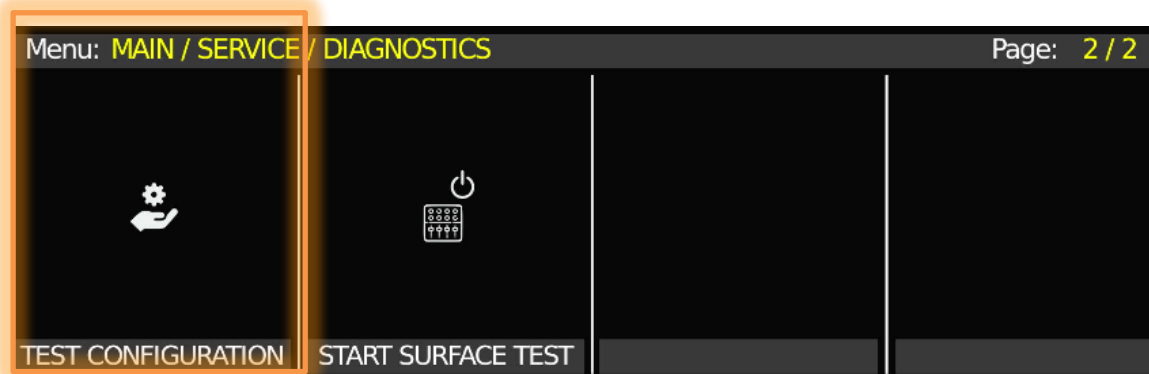
- **SINGLE PSU**
- **REDUNDANT PSU**

When **REDUNDANT PSU** is selected, the mixer monitors the availability and operating condition of both power-supply units, ensuring redundancy supervision is active. When **SINGLE PSU** is selected, the system

operates with a single power-supply unit and monitoring is limited to that single source, without redundancy control.

TEST CONFIGURATION:

The **TEST CONFIGURATION** menu, located under **SERVICE → DIAGNOSTIC**, defines which internal diagnostic tests are performed when **START SURFACE TEST** is executed.

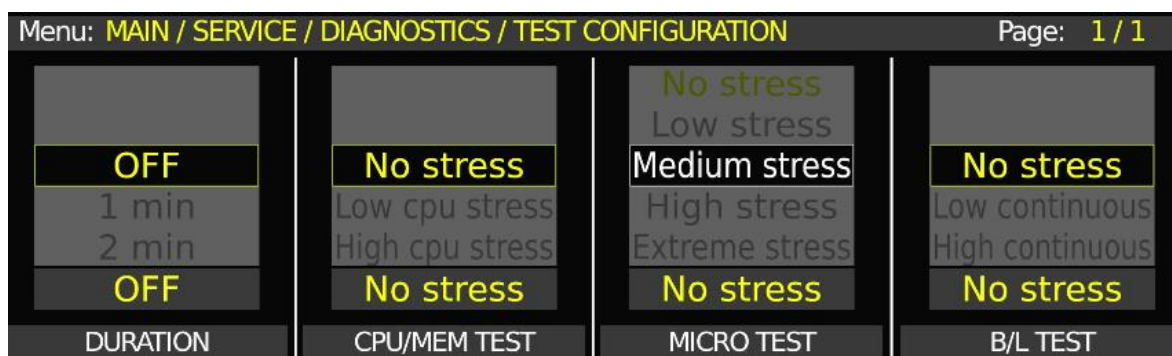


This section is used to configure the duration of the procedure and enable or disable the individual test categories. The TEST CONFIGURATION page includes:

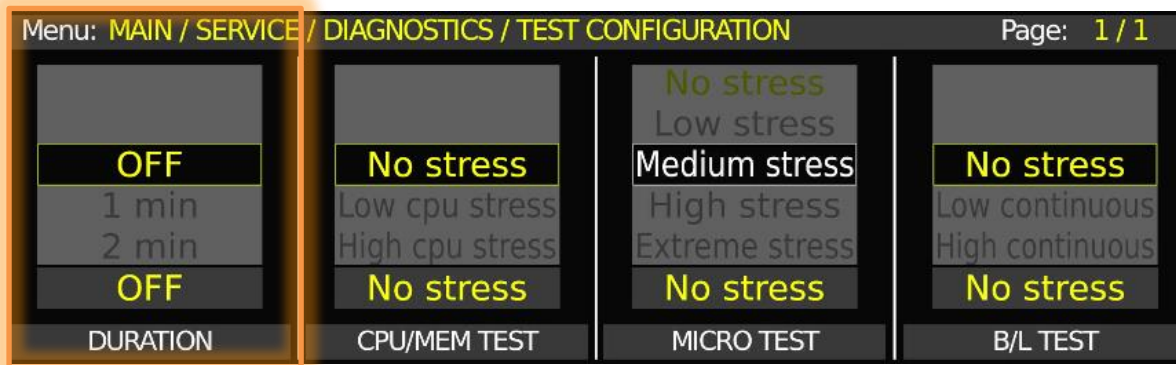
- **DURATION**
- **CPU/MEM TEST**
- **MICRO TEST**
- **B/L TEST**

Selecting these parameters does not immediately start the diagnostic procedure. The configured tests begin only when **START SURFACE TEST** is activated.

Important: perform the diagnostic test only when the console is **not On Air**. The procedure may place the control surface under diagnostic load and temporarily prevent normal operation of buttons, LEDs, displays or other controls.



DURATION:



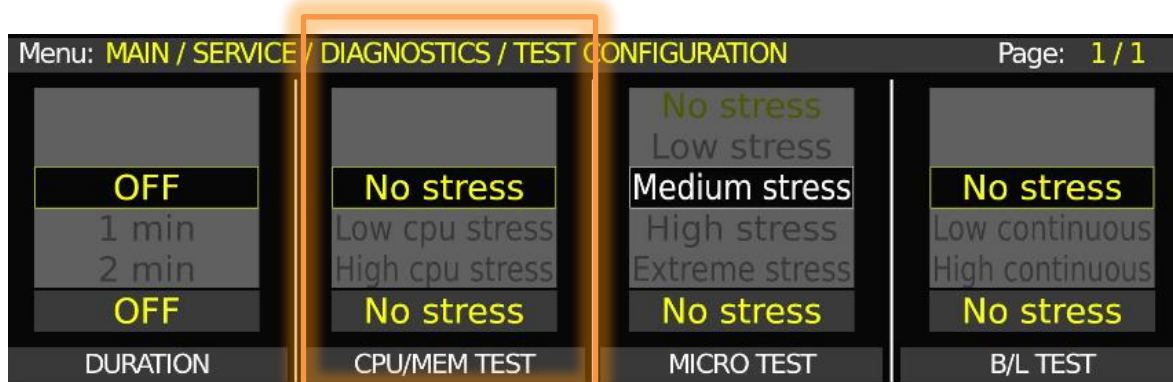
DURATION defines how long the selected diagnostic tests must run.

Selectable setting:

- A time interval ranging from **a few minutes up to several days**, depending on the selected configuration

The duration can therefore be adjusted from short verification cycles to extended endurance tests. **DURATION** applies to the tests enabled under **CPU/MEM TEST**, **MICRO TEST** and **B/L TEST**. **Important:** **DURATION** defines only the execution time of the diagnostic procedure. It does not affect audio routing, system configuration, logs retention or normal console operation.

CPU/MEM TEST:

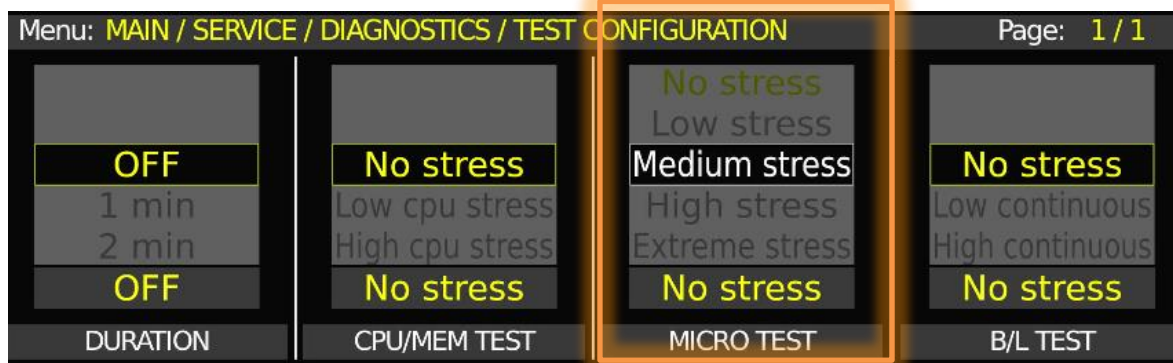


CPU/MEM TEST enables or disables the diagnostic stress test of the console's main processor and memory resources.

The selectable options are:

- **No stress:** no CPU or memory load is applied; only minimal monitoring is performed
- **Low CPU stress:** applies a light processing load to the CPU
- **High CPU stress:** applies an intensive processing load to the CPU
- **Low CPU + MEM stress:** applies a combined light load to CPU and memory
- **High CPU + MEM stress:** applies a combined intensive load to CPU and memory

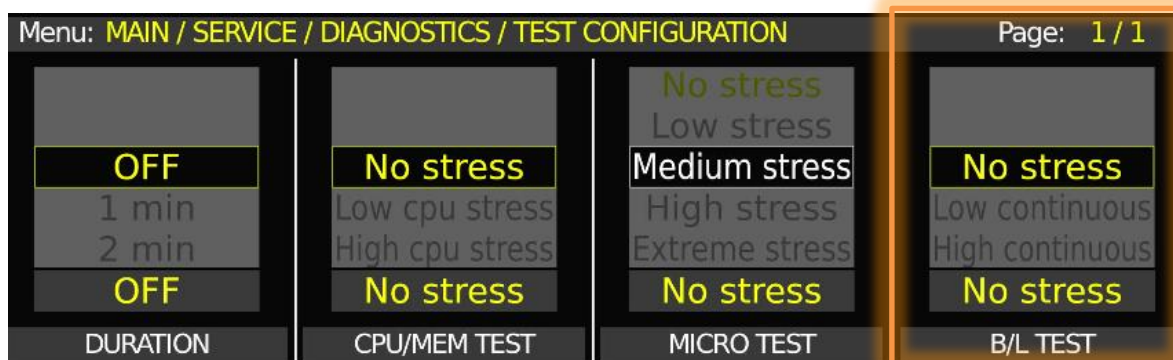
Important: CPU/MEM TEST is a diagnostic stress tool. It does not modify system configuration or repair hardware faults.

MICRO TEST:

MICRO TEST enables or disables the stress test of the internal microcontrollers installed in the control surface and related modules. The selectable options are:

- **No stress:** basic communication check without load
- **Low stress:** light communication and response verification
- **Medium stress:** moderate load on communication channels and module responses
- **High stress:** intensive communication and response verification
- **Extreme stress:** maximum load condition for extended reliability testing

Important: MICRO TEST refers exclusively to internal control-surface microcontrollers and does not involve audio signal processing or microphone inputs.

B/L TEST:

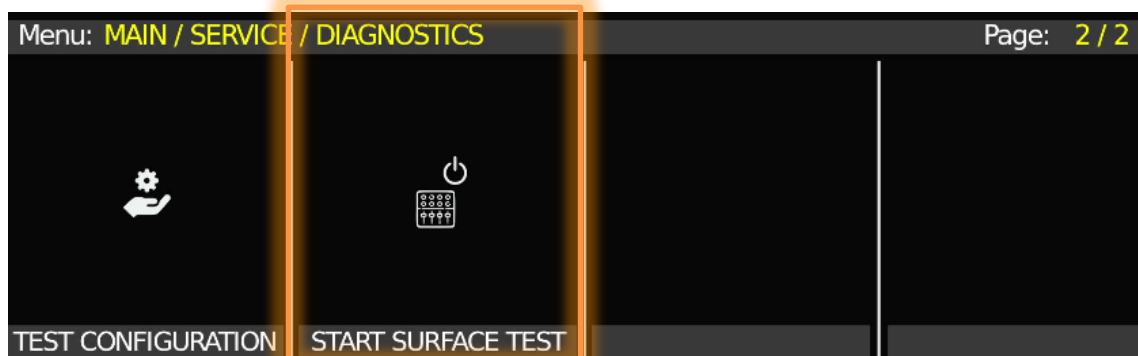
B/L TEST enables or disables the stress test of surface **buttons and lighting (LED) systems**.

The selectable options are:

- **No stress:** no continuous or pulsed activation is applied
- **Low continuous:** continuous low-intensity activation of buttons and LEDs
- **High continuous:** continuous high-intensity activation of buttons and LEDs
- **Low impulsive:** intermittent low-frequency pulsed activation
- **High impulsive:** high-frequency pulsed activation with rapid state changes

During the test, controls may continuously change state or be repeatedly activated according to the selected stress profile. **Important:** while B/L TEST is active, all visual feedback is controlled by the diagnostic system and does not reflect On Air status, routing, PFL or channel activity.

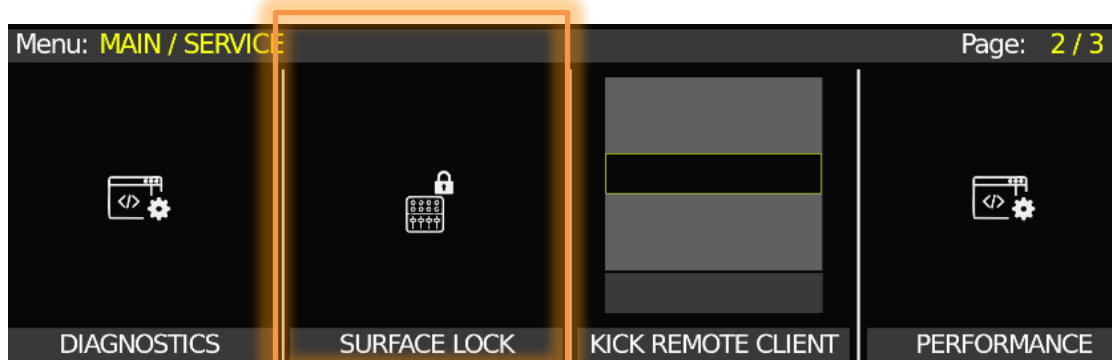
START SURFACE TEST:



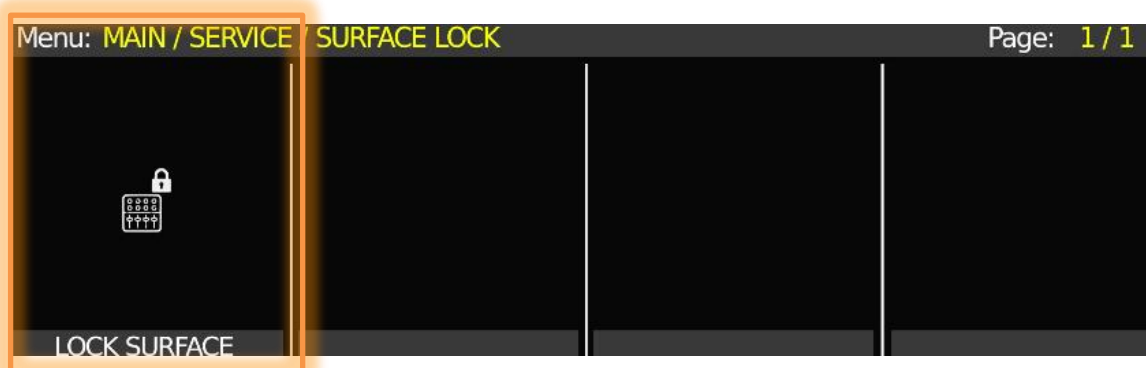
START SURFACE TEST begins the interactive test of the OXYGEN 3000 Plus physical control surface using the profile selected under TEST CONFIGURATION.

6. SURFACE LOCK

The **SURFACE LOCK** function, located under **SERVICE**, temporarily locks the physical control surface of the OXYGEN 3000 Plus. It is designed to prevent accidental or unauthorised changes to the console during situations.



When SURFACE LOCK is active, the console ignores normal operations performed through physical controls. The current audio configuration remains stored and the console continues operating with the settings that were active when the surface was locked.



Unlocking the surface

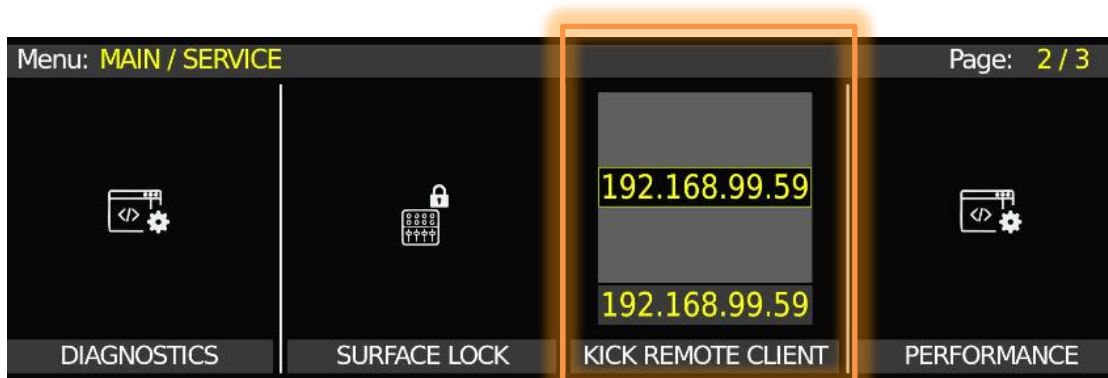
To unlock the OXYGEN 3000 Plus control surface:

1. Locate the **four rotary menu encoders** associated with the central display.
2. Press all four encoders simultaneously.
3. Continue holding them for approximately **five seconds**.
4. Release them when the console indicates that the surface has been unlocked.
5. Verify that the faders, buttons and rotary controls respond normally.

The official procedure requires the operator to **press and hold all four menu-control knobs together for five seconds**.

7. KICK REMOTE CLIENT

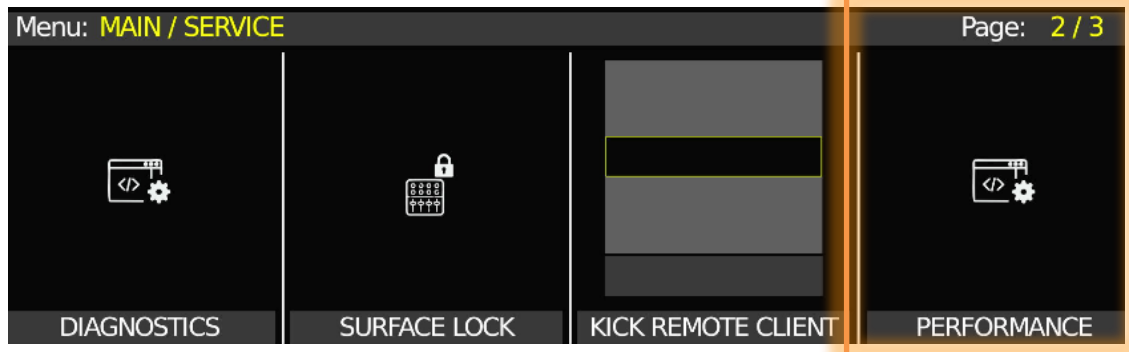
KICK REMOTE CLIENT disconnects a computer that is currently controlling the mixer through **OXYGEN Remoter**. The **REMOTE CLIENT** is displayed as a sequence of numbers. To disconnect it, select **KICK REMOTE CLIENT** and press the corresponding **KNOB**. **KICK REMOTE CLIENT** disconnects the selected session but does not uninstall OXYGEN Remoter, change the console configuration or permanently block that computer from reconnecting.



Security note: if the disconnected client must not reconnect, also change the applicable access credentials and verify the network-access configuration.

8. PERFORMANCE

The **PERFORMANCE** menu, located under **SERVICE**, configures the operating frequency of selected background and graphical processes of the OXYGEN 3000 Plus. It can be accessed through: **MAIN → SERVICE → PERFORMANCE**

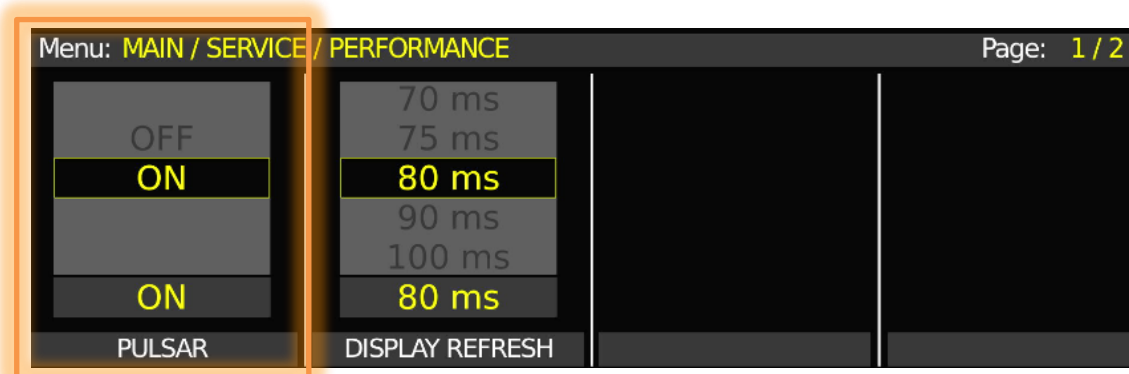


These parameters determine how frequently the console updates internal status information and refreshes the data shown on its displays. The **PERFORMANCE** section includes:

- **PULSAR**
- **DISPLAY REFRESH**

Important: **PERFORMANCE** parameters affect system and graphical-update behaviour. They do not configure audio routing, audio sample rate, input gain or DSP processing.

PULSAR:



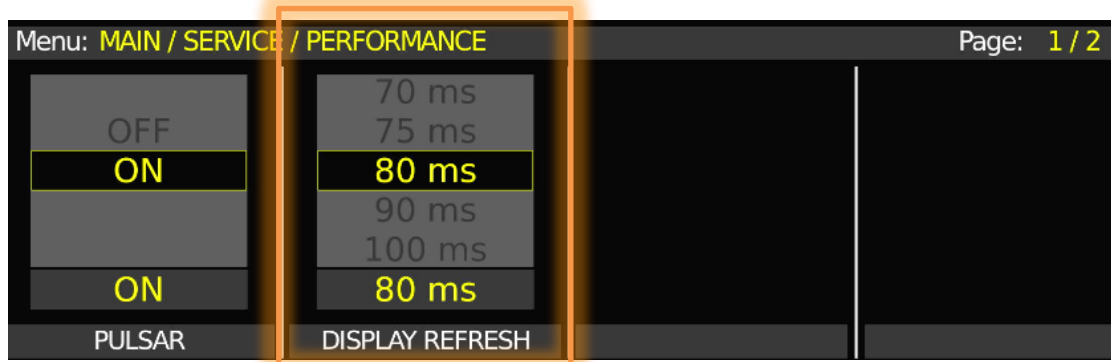
PULSAR enables or disables the console's internal background pulse or polling process. This process is responsible for periodically exchanging and updating status information between the main software, control surface and associated system services. It helps maintain synchronisation of information.

Selectable setting:

- **ON:** When **PULSAR** is set to **ON**, the internal polling process is active. This provides: Continuous update of surface and system status - Real-time synchronisation between software components - Immediate propagation of control changes - Correct and up-to-date graphical feedback on displays This is the **normal operating condition** of the console.
- **OFF:** When **PULSAR** is set to **OFF**, the internal polling process is disabled. **Important:** **PULSAR OFF** is intended only for diagnostic or technical procedures. It is not recommended for normal On Air

operation. **Important:** PULSAR does not change the console's audio clock, sample rate or audio-processing latency. It only affects the update of internal status and control information.

DISPLAY REFRESH:



DISPLAY REFRESH defines how frequently the graphical information shown on the OXYGEN 3000 Plus displays is updated. **Selectable setting:**

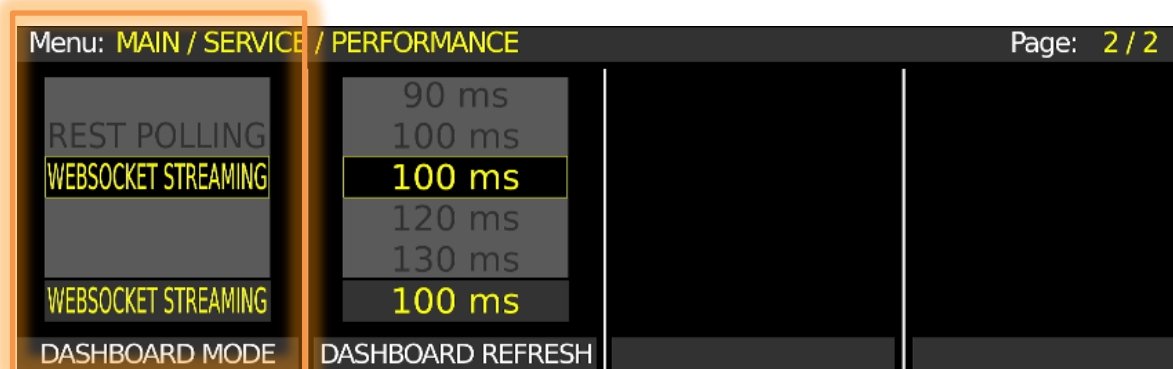
- **One of the display-refresh intervals or performance values available in the installed firmware**

The available selections can generally be interpreted as:

- **Faster refresh / shorter interval:** redraws the displayed information more frequently.
- **Default or nominal refresh:** provides the standard balance between graphical smoothness and system workload.
- **Slower refresh / longer interval:** reduces the frequency of graphical updates.

Important: DISPLAY REFRESH controls how frequently information is redrawn by the console software. It does not change the physical panel frequency, HDMI video format or audio signal.

DASHBOARD MODE:



DASHBOARD MODE selects the method used to transfer console status information to the Dashboard.

The selectable options are:

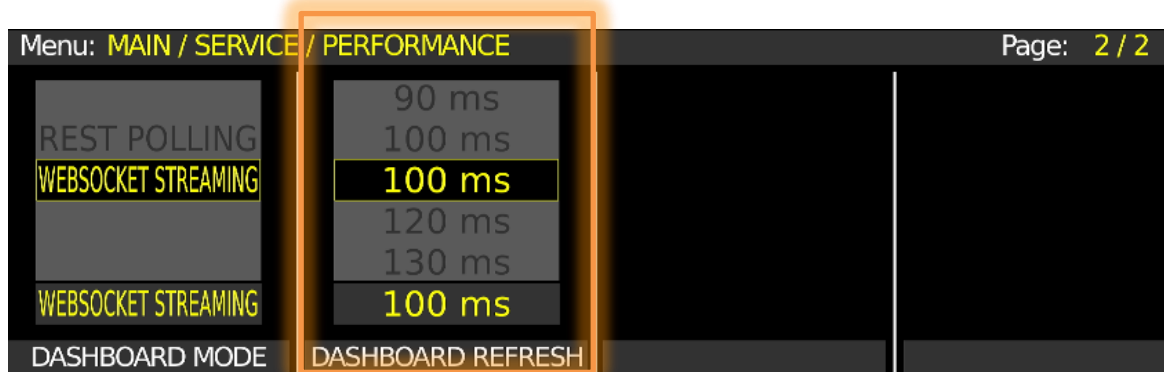
- **REST POLLING:** When DASHBOARD MODE is set to **REST POLLING**, the Dashboard obtains updated console information by sending repeated requests to the console. In this mode **DASHBOARD REFRESH** determines the request frequency. **REST POLLING may be suitable** when: Compatibility with standard web communication is required - The network or browser environment does not

support persistent streaming connections - The Dashboard is used for general monitoring - A controlled and predictable request interval is preferred.

- **WEBSOCKET STREAMING:** When DASHBOARD MODE is set to **WEBSOCKET STREAMING**, the Dashboard uses a persistent WebSocket connection to receive status updates from the console. In this mode: The Dashboard establishes a continuous communication channel - The console can send updates without waiting for a new polling request - Status changes can become visible with lower communication delay. **WEBSOCKET STREAMING may be suitable** when: Fast Dashboard response is required - The Dashboard is used for live operational monitoring.

These options do not select the Dashboard layout. Display layouts, logos and HDMI presentation are configured through the corresponding display settings.

DASHBOARD REFRESH:



DASHBOARD REFRESH determines how frequently the Dashboard requests or redraws updated console information, primarily when REST POLLING is selected. The available settings are:

- **OFF:** When DASHBOARD REFRESH is set to **OFF** in **REST POLLING** mode, periodic Dashboard polling is disabled. OFF may be suitable when:
 - The Dashboard is not required for continuous monitoring - Network traffic must be minimized - A static status view is acceptable - Diagnostic testing is being performed
- Any interval **from 25 ms to 2000 ms:**

9. DETECT HDMI DISPLAY



DETECT HDMI DISPLAY is a system command used to detect or re-detect an external monitor connected to the **HDMI output** of the OXYGEN 3000 Plus.

ATTENTION: If an external monitor is connected to the **Oxygen 3000 Plus** while the console is powered off, it will be automatically detected when the system is switched on. If the monitor is connected while the console is already operating, it may not be detected automatically. The **DETECT HDMI DISPLAY** function allows the console to detect and connect to the monitor without requiring the mixer to be powered off or restarted.

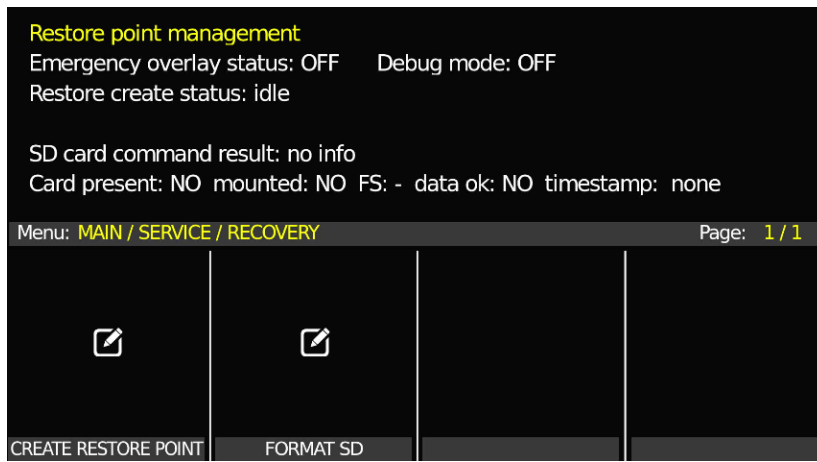
10. RECOVERY:

RECOVERY, located under **SERVICE**, is a maintenance function used to restore the OXYGEN 3000 Plus to a normal operating state following a serious software fault, an incomplete startup or another abnormal system condition. It can be accessed through: **MAIN → SERVICE → RECOVERY**



Depending on the installed firmware, **RECOVERY** may reinitialise the console's main software components, restart internal services and reload the software environment required for normal operation. **Important:** **RECOVERY** is an advanced service function. It should not be used as a routine restart command or as a general troubleshooting attempt.

Warning: never switch off the console or disconnect its power supply while **RECOVERY** is active. Interrupting the procedure may leave the operating software incomplete or prevent the mixer from restarting correctly.

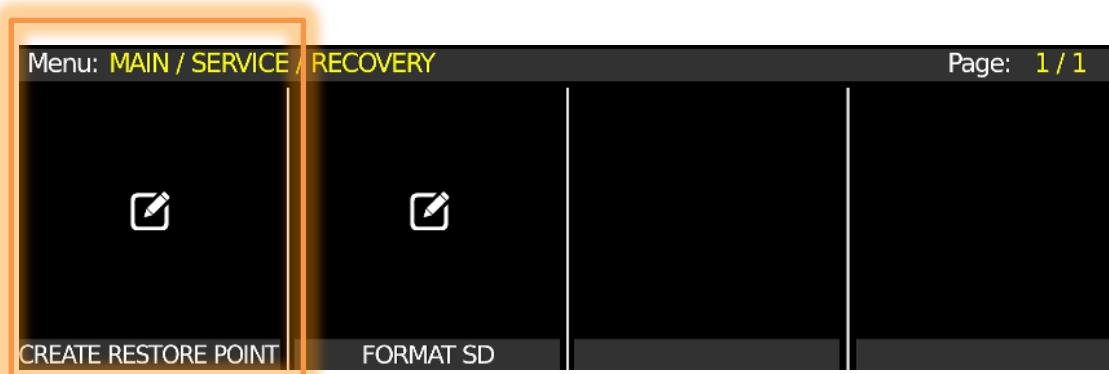


The section includes two commands:

- **CREATE RESTORE POINT**
- **FORMAT SD**

These commands allow the operator to create a recoverable system state and manage the SD storage used to retain recovery data.

CREATE RESTORE POINT:



CREATE RESTORE POINT saves the current system state to the console's internal SD memory, creating a reference that can be used if the system must later be returned to a previously working condition. During creation, the console may display a progress indication. The operation can require some time, depending on the amount of information being stored. **Important:** do not switch off or restart the mixer while the restore point is being created. An interrupted procedure may produce incomplete or unusable recovery data.

FORMAT SD:



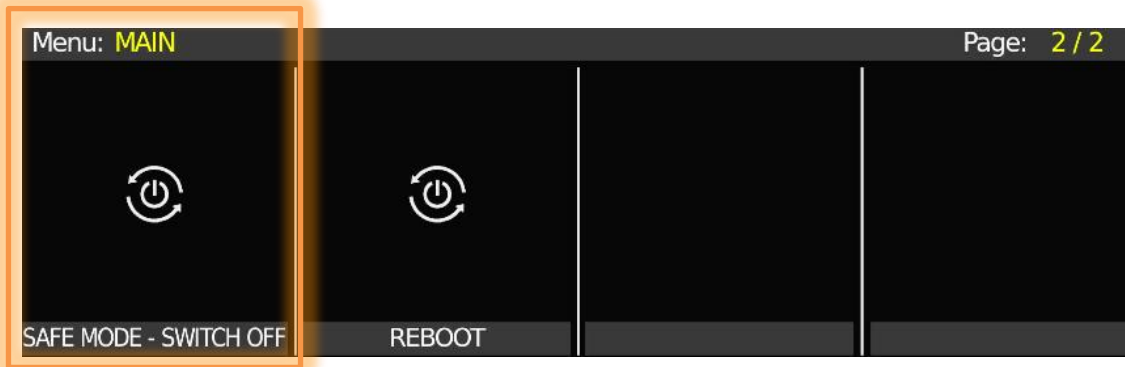
FORMAT SD erases and reinitialises the SD storage area used by the console. Formatting creates a new file system on the SD memory, removing the data currently stored in the affected storage area.

Warning: FORMAT SD permanently deletes the restore points and other files stored in the affected SD area. Save all required configurations and diagnostic information to an external USB device before confirming the operation.

SAFE MODE – SWITCH OFF

The **SAFE MODE – SWITCH OFF** command, located under **MAIN**, performs a controlled shutdown of the OXYGEN 3000 Plus before the console is switched off or disconnected from its power supply. It can be accessed through: **MAIN → SAFE MODE – SWITCH OFF**

In this context, **SAFE MODE** does not indicate a diagnostic startup mode with reduced functions. It identifies the protected state reached after the console has correctly stopped its internal software and completed the operations required for safe power removal.



During this procedure, the user must wait until the console has reached its final safe state.

Warning: do not operate the power switch, disconnect the power cable or remove the internal SD card while the shutdown procedure is still in progress.

Once the console reports that shutdown is complete, or when its displays and control surface have entered the final inactive state, the external power supply can be switched off.

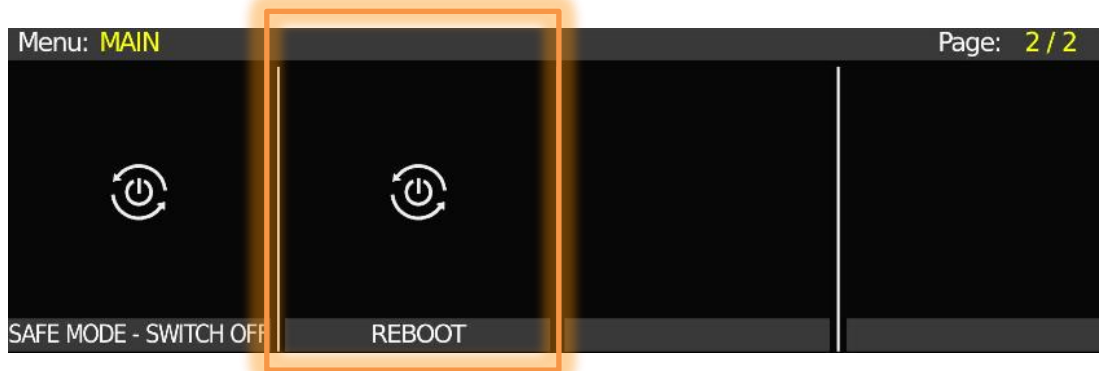
It's now safe to turn off
your mixer

REBOOT

The **REBOOT** command, located under **MAIN**, performs a controlled restart of the OXYGEN 3000 Plus.

It can be accessed through: **MAIN → REBOOT**

When REBOOT is executed, the console closes its active software processes and automatically restarts the operating system, control surface and associated services. Unlike **SAFE MODE – SWITCH OFF**, the console does not remain in a shutdown state and the power supply does not need to be manually switched off.

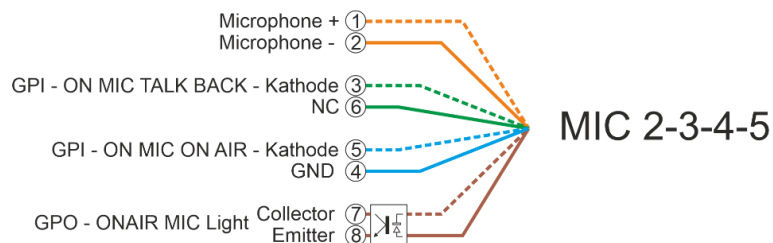
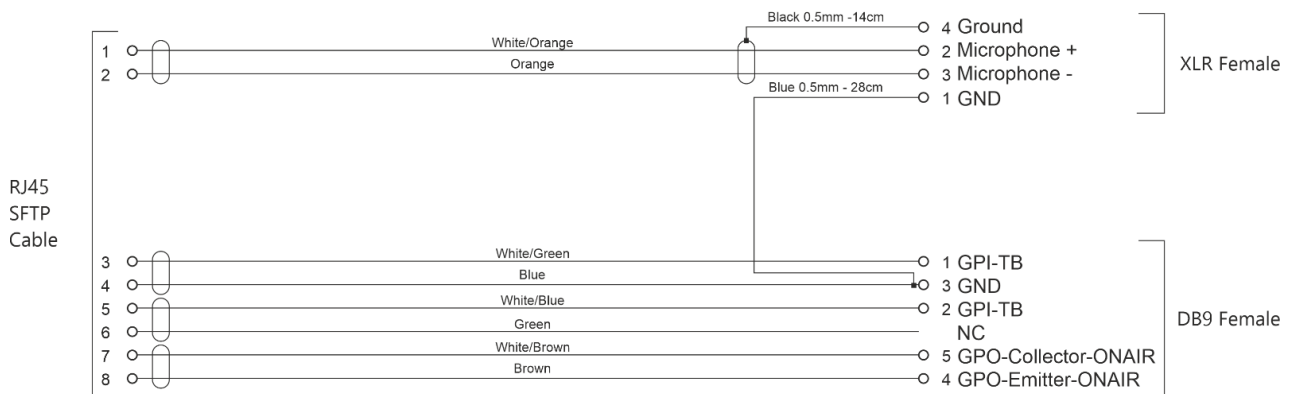
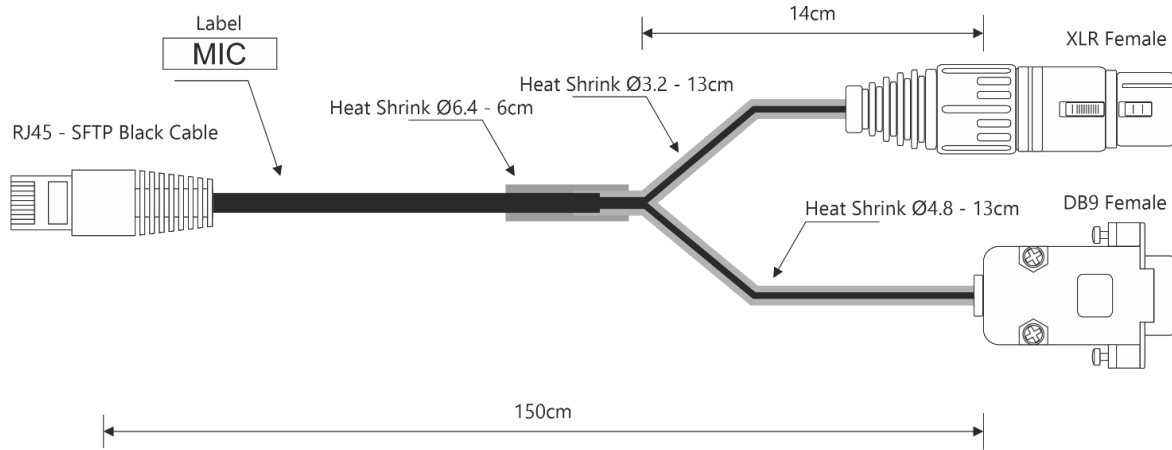


REBOOT may be used to:

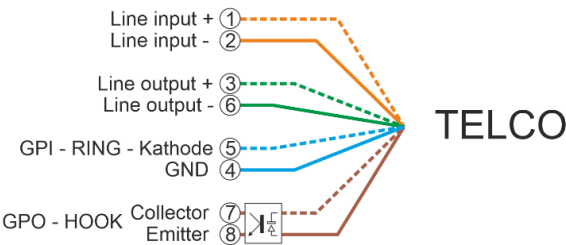
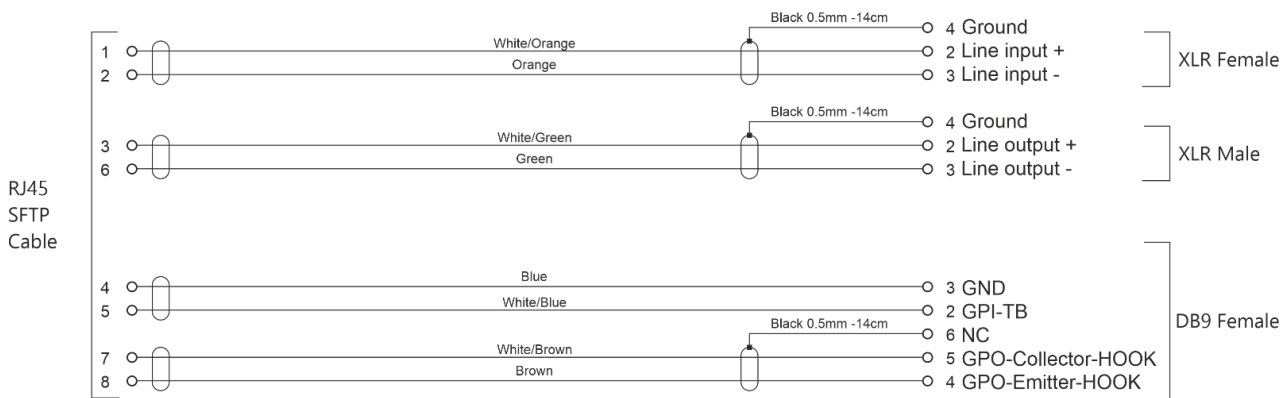
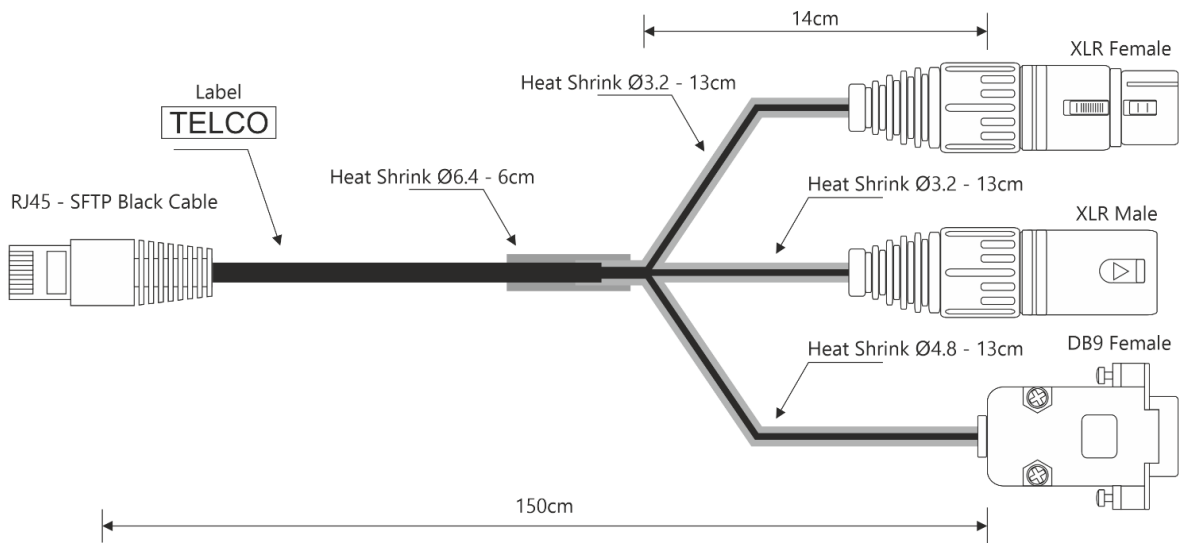
- Restart the console after changing system parameters
- Reload the current firmware and stored configuration
- Reinitialise the control surface
- Restart network and remote-control services
- Detect installed hardware and connected peripherals again
- Recover from a temporary software or communication problem
- Restore normal display or control behaviour after an anomaly
- Complete maintenance procedures that require a system restart

Important: REBOOT restarts the entire console. Normal audio and control operation may be interrupted until the startup procedure has been completed. **Important:** REBOOT is not a substitute for the configuration **SAVE**, backup or **CREATE RESTORE POINT** functions. Save or export important settings before performing maintenance.

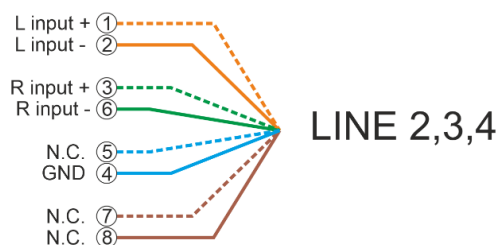
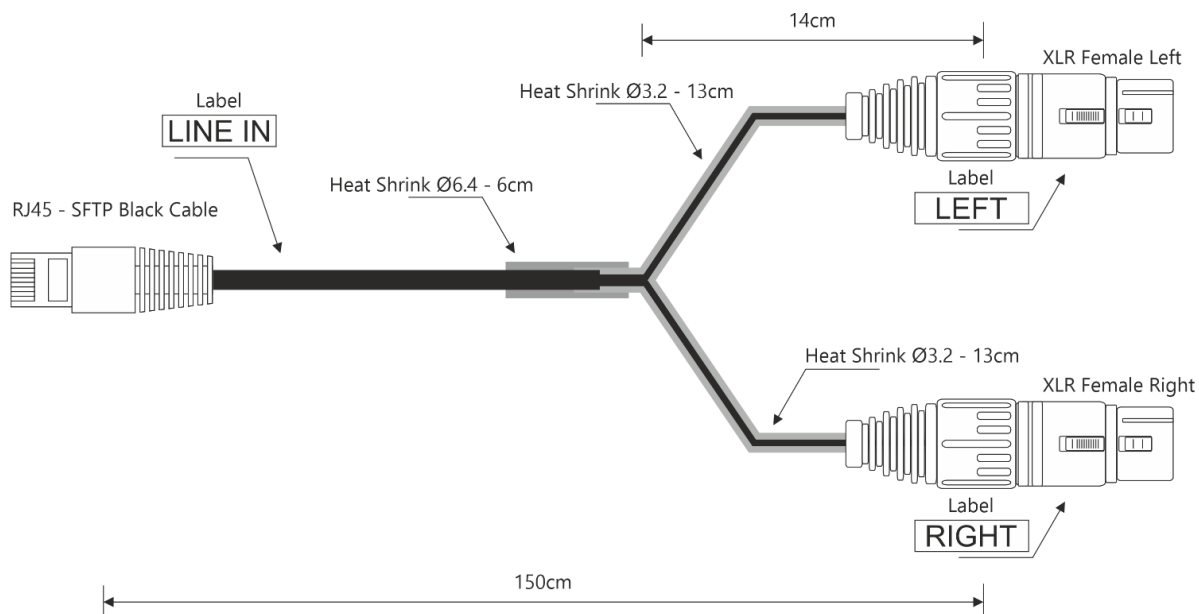
+ 187-RJ45-MIC



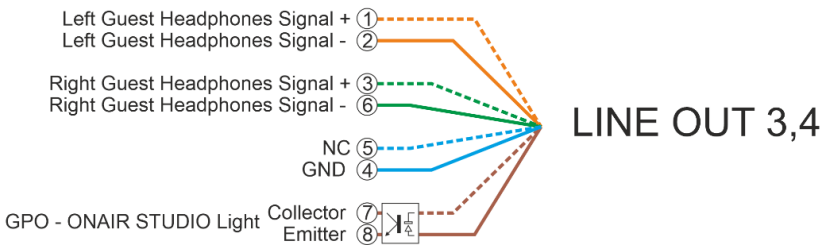
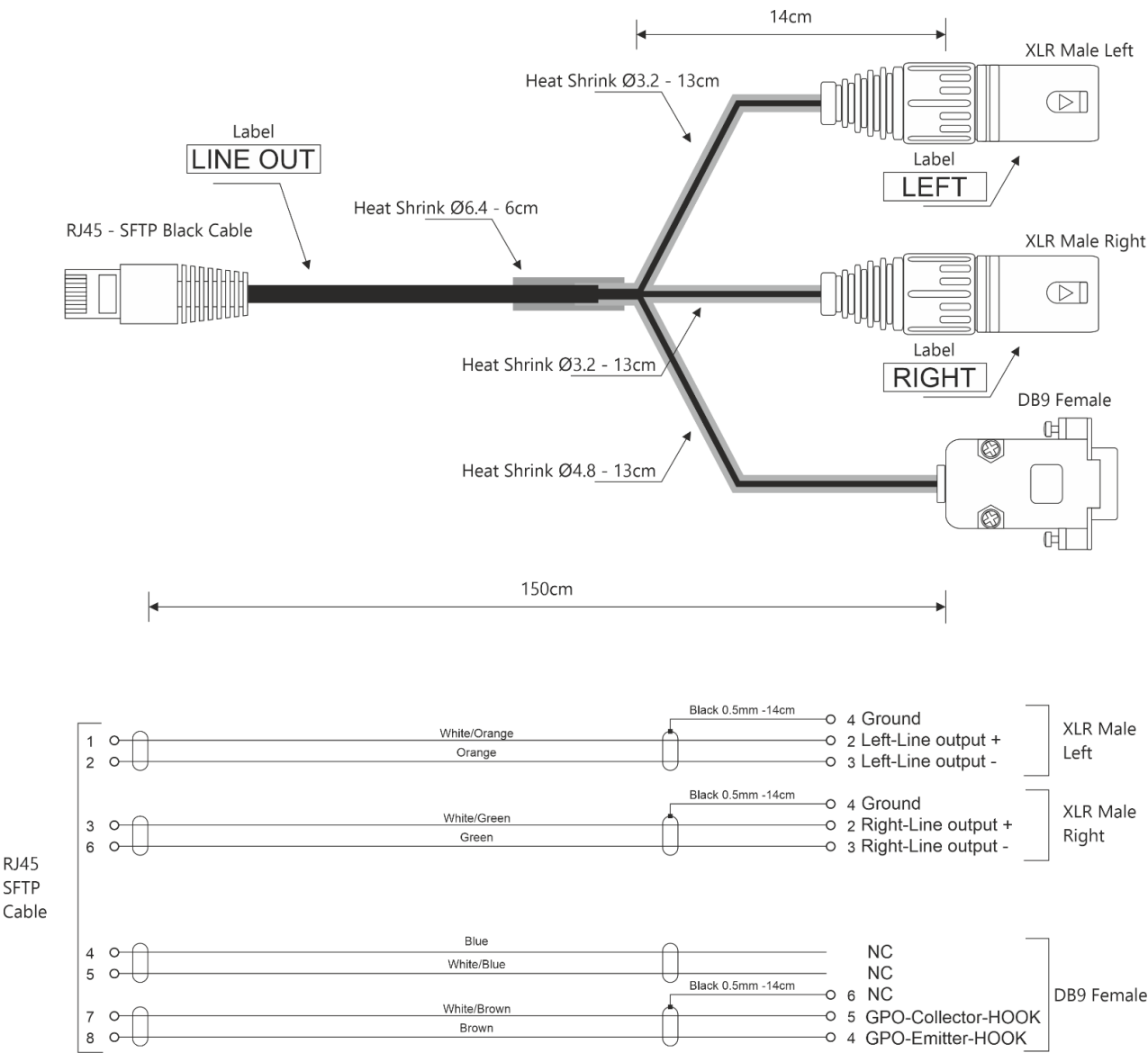
+ 188-RJ45-TELCO



+ 189-RJ45-LINE-IN



+ 190-RJ45-LINE-OUT



TECH SPECS

Analog Balanced Microphone Inputs

Connector RJ45 & XLR Balanced – EMI Suppressed

Input Impedance 2,4 K Ω

Nominal Input Level (sensitivity) -9/-66 dBu

Max Input Level (clipping point) +9 dBu

A/D conversion 24-bit / 48 kHz

Signal To Noise Ratio (referred as peak level) >110 dB

THD+N <0,01%

Analog Gain Adjustable +0 ÷ +60 dB (3dB step)

Phantom Power +48V

Analog Balanced Stereo Inputs

Connector XLR & RJ45 Balanced – EMI Suppressed

Input Impedance 10 K Ω

Nominal Input Level (sensitivity) 0 dBu

Max Input Level (clipping point) +18 dBu

A/D conversion 24-bit / 48 kHz

Frequency response +/-0,5 dB from 20 Hz to 20 kHz

Signal To Noise Ratio (referred to as peak level) >100 dB

Stereo Separation (referred to as peak level) >90 dB

THD+N <0,002 %

Analog Balanced Telco Input

Connector RJ45 Balanced – EMI Suppressed

Input Impedance 10 K Ω

Nominal Input Level (sensitivity) 0 dBu

Max Input Level (clipping point) +18 dBu

A/D conversion 24-bit / 48 kHz

Signal To Noise Ratio (referred to as peak level) >110 dB

THD+N <0,002%

Digital Input

Connector Balanced on 1 XLR – EMI Suppressed

Input Impedance 110 K Ω

Standard AES3

Audio Sample Rate 32/44.1/48/96/192 KHz with SRC

Resolution 24 bit

Dynamic Range (Converter Values) 124 dB

Analog Balanced Stereo Outputs

Connector RJ45 Balanced – EMI Suppressed

Output Impedance 23 Ω , nominal 600 Ω

Nominal Output Level 0 dBu

Max Output Level (clipping point) +18 dBu

D/A conversion 24 bit / 48 KHz

Signal To Noise Ratio (referred to peak level) >110 dB

Stereo Separation (referred to peak level) >90 dB

THD+N <0,002 %

Dante I/O

Audio Buffering Up to 2000 samples per channel

Audio Flows In/Out Up to 32x32 simultaneous streams

Audio Transport Formats Dante Audio over IP, AES67 RTP, SMPTE ST2110-30 RTP (enrolled devices)

Clock High quality, low jitter onboard SiLabs

Clocking Onboard word clock or external word clock

Control Interfaces SPI Master and Slave; GPIO; I2C

Digital Audio Formats TDM, I2S

Ethernet Standard RGMII/MII interface for Ethernet PHY or switch chip

Form Factor Card edge module. 4.5cm x 6cm (1.75" x 2.4")

FPGA High performance Xilinx Spartan6 FPGA

Microprocessor Soft-core Microblaze processor

Network RGMII/MII

Dante I/O

Physical Connector Mini-PCI

Power 3.3VDC @ 2W max

Sample Bit Depth 16, 24 or 32 bits per sample

Sample Rates (16x16 Versions) 44.1/48/88.2/96 kHz

Sample Rates (32x32 Versions) 44.1/48/88.2/96 kHz

USB Audio Digital I/O

Connector USB Type B – EMI Suppressed

Playback And Recording Sample Rate SRC 44.1-48 KHz

Resolution 16 bit

Available Stereo Channels 2 stereo USB - 2 Inputs/Output on each Audio Board

Analog Balanced Talkbox Outputs

Connector RJ45 balanced – EMI Suppressed

Output Impedance 100ohm, nominal 600ohm

Nominal Output Level 0dBu

Max Output Level (clipping point) +14dBu

D/A conversion 24 bit / 48 Khz

Signal To Noise Ratio (referred to peak level) >110dB

Stereo Separation (referred to peak level) >90dB

THD+N <0,05%

Analog Balanced Telco Output

Connector RJ45 Balanced – EMI Suppressed

Output Impedance 23 Ω, nominal 600 Ω

Nominal Output Level 0 dBu

Max Output Level (clipping point) +18 dBu

D/A conversion 24 bit / 48 Khz

Signal To Noise Ratio (referred to peak level) >110 dB

THD+N <0,002 %

Digital Output

Connector Balanced on 1 XLR – EMI Suppressed

Input Impedance 110 Ω

Standard AES3

Audio Sample Rate 32/44.1/48/96/192 KHz

Resolution 24 bit

Dynamic Range (Converter Values) 124 dB

PSTN Interface

Connector RJ11

Transhybrid loss >20 dB

System

Audio Core Analog Devices ADAU1452 32bit 294 MHz fixed point DSP

Audio CODECs Cirrus CS42448 24 bit/192 kHz

LAN Connection RJ45 - 1 Gbit

Nominal Delay (analog input to analog output) 0,7 ms

GPIO Inputs/Outputs 4 GPIO on DB9; 4 GPI/2 GPO on Mic2 & Mic3 RJ45; 2 GPO on Out3 & Out4 RJ45; 1 GPI/1 GPO on Telco RJ45

Communication Port 2xUSB type-A, 2xUSB type-B , 1xLAN, 1xHDMI

Operating Temperature 0°C ÷ 40°C

Dimensions

8 Faders Dimensions (W; H; D) 584 mm; 113,5 mm; 495,5 mm

12 Faders Dimensions (W; H; D) 764 mm; 113,5 mm; 495,5 mm

16 Faders Dimensions (W; H; D) 944 mm; 113,5 mm; 495,5 mm

Weight 15 to 25 Kg

PSU

Type External Universal Switching Power Supply 12V with 4 poles XLR connections - Optional Redundant

Power Supply 90-264 VAC / 50-60 Hz / 120 W

WEEE DIRECTIVE – INFORMATIVA RAEE



In line with EU Directive 2002/96/EC for waste electrical and electronic equipment (WEEE), this electrical product must not be disposed of as unsorted municipal waste. Please dispose of this product by returning it to the point of sale or to your local municipal collection point for recycling.

In Übereinstimmung mit der Richtlinie 2002/96/EG des Europäischen Parlaments

und des Rates über Elektro- und Elektronik-Altgeräte (WEEE) darf dieses Elektrogerät nicht im normalen Hausmüll oder dem Gelben Sack entsorgt werden. Wenn Sie dieses Produkt entsorgen möchten, bringen Sie es bitte zur Verkaufsstelle zurück oder zum Recycling-Sammelpunkt Ihrer Gemeinde.

Conformément à la Directive 2002/96/EC sur les déchets d'équipements électriques et électroniques (DEEE), ce produit électrique ne doit en aucun cas être mis au rebut sous forme de déchet municipal non trié. Veuillez vous débarrasser de ce produit en le renvoyant à son point de vente ou au point de ramassage local dans votre municipalité, à des fins de recyclage.

In navolging van richtlijn 2002/96/EG van het Europees Parlement en de Raad betreffende afgedankte elektrische en elektronische apparatuur (AEEA) mag dit elektrische product niet als ongescheiden huisvuil worden weggedaan. Breng dit product terug naar de plaats van aankoop of naar het gemeentelijke afvalinzamelingspunt voor recycling.

In ottemperanza alla Direttiva UE 2002/96/EC sui rifiuti di apparecchiature elettriche ed elettroniche (RAEE), questo prodotto elettrico non deve essere smaltito come rifiuto municipale misto. Si prega di smaltire il prodotto riportandolo al punto vendita o al punto di raccolta municipale locale per un opportuno riciclaggio.

De conformidad con la Directiva 2002/96/CE de la UE sobre residuos de aparatos eléctricos y electrónicos (RAEE), este producto eléctrico no puede desecharse con el resto de residuos no clasificados. Deshágase de este producto devolviéndolo al punto de venta o a un punto de recogida municipal para su reciclaje.

I henhold til EU-direktiv 2002/96/EF om affald af elektrisk og elektronisk udstyr (WEEE) må dette udstyr ikke bortskaffes som usorteret husholdningsaffald. Bortskaf dette produkt ved at returnere det til salgsstedet eller til det lokale indsamlingssted, så det kan genbruges.

I linje med EU-direktiv 2002/96/EG om avfall som utgörs av eller innehåller elektriska eller elektroniska produkter (WEEE) får denna elektriska produkt inte bortskaffas som osorterat kommunalt avfall. Bortskaffa den i stället genom att lämna in den på försäljningsstället eller din lokala återvinningsstation.

EU:n sähkö- ja elektroniikkalaiteromudirektiivin (2002/96/EY) mukaisesti tätä elektroniikkalaitetta ei saa laittaa jätteen sekaan. Hävitä laite palauttamalla se ostopaikkaan tai viemällä se elektroniikkaromun keräyspisteeseen.

De acordo com a Directiva Europeia 2002/96/EC sobre resíduos sólidos de equipamento eléctrico e electrónico (WEEE), este produto eléctrico não pode ser deixado fora juntamente com o lixo municipal indiferenciado. Por favor, no final da vida útil deste produto, devolva-o ao estabelecimento de aquisição, ou entregue no local de recolha apropriado para reciclagem designado pelo seu município.

V souladu se smrnici EU . 2002/96/ES o odpadních elektrických a elektronických zařízeních (OEEZ) se tento elektrický výrobek nesmí likvidovat jako neřádný komunální odpad. Při likvidaci výrobek vraťte prodejci nebo ho odevzdejte k recyklaci do komunálního sborného zařízení.

Vastavalt EL direktiivile 2002/96/EÜ, mis käsitleb elektri- ja elektroonikaseadmete jäätmeid (WEEE), ei või antud toodet visata majapidamisjäätmete hulka. Palun tagastage antud toode taaskasutamise eesmärgil müügipunkti või kohaliku piirkonna jäätmekogumise punkti.

V súlade so smernicou 2002/96/ES o odpade z elektrických a elektronických zariadení (OEEZ) sa toto elektrické zariadenie nesmie odstranovať ako netriedený komunálny odpad. Výrobok odstráňte jeho vrátením v mieste nákupu alebo odovzdaním v miestnom zbernom zariadení na recyklovanie.

V súlade so smernicou 2002/96/ES o odpade z elektrických a elektronických zariadení (OEEZ) sa toto elektrické zariadenie nesmie odstranovať ako netriedený komunálny odpad. Výrobok odstráňte jeho vrátením v mieste nákupu alebo odovzdaním v miestnom zbernom zariadení na recyklovanie.

WARRANTY

The manufacturer offers a one-year warranty ex-works. Do not open the equipment. Any breaking of the seals will result in the forfeiture of the same. The manufacturer is not liable for damages of any kind arising from, or in connection with, the use of the wrong product.