Operating Manual

OPTIMOD-PCn 1600/1601/1602

Digital Audio Processor Software
for Intel/Windows PCs

Manual for Software Version 0.9.18

orban

IMPORTANT NOTE: Refer to the unit’s rear panel for your Model #.

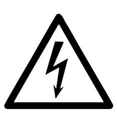
Model Number:	Description:
1600PCn	OPTIMOD-PCn Digital Audio Processor and Loudness Controller Software running natively on an Intel®-based Windows® computer. Optionally includes Modulation Index StreamS streaming software for netcasting.
1601PCn	Lower-cost version of software with certain features omitted. See *1601PCn Features* on page 1-12.
1602PCn	Lowest-cost version of software with certain features omitted. See *1602PCn Features* on page 1-12.

CAUTION

RISK OF ELECTRIC SHOCK
DO NOT OPEN

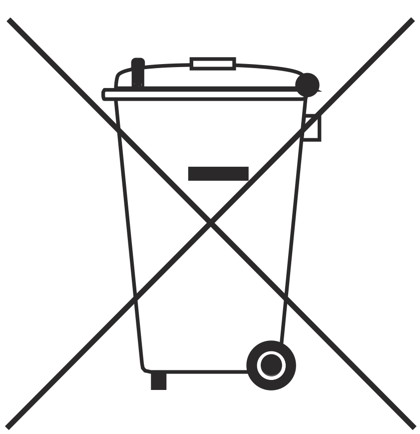
CAUTION: TO REDUCE THE RISK OF ELECTRICAL SHOCK, DO NOT REMOVE COVER (OR BACK).
NO USER SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

WARNING: TO REDUCE THE RISK OF FIRE OR ELECTRICAL SHOCK,
DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.



This symbol, wherever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure — voltage that may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Read the manual.



In accordance to the WEEE (waste electrical and electronic equipment) directive of the European Parliament, this product must not be discarded into the municipal waste stream in any of the Member States. This product may be sent back to your Orban dealer at end of life where it will be reused or recycled at no cost to you.

If this product is discarded into an approved municipal WEEE collection site or turned over to an approved WEEE recycler at end of life, your Orban dealer must be notified and supplied with model, serial number and the name and location of site/facility.

Please contact your dealer for further assistance.

www.orban.com

PLEASE READ BEFORE PROCEEDING!

Manual

Please review the Manual, especially the installation section, before installing the unit in your computer.

Trial Period Precautions

If your unit has been provided on a trial basis:

You should observe the following precautions to avoid reconditioning charges in case you later wish to return the unit to your dealer.

(1) Note the packing technique and save all packing materials. It is not wise to ship in other than the factory carton. (Replacements cost $35.00).

(2) Avoid scratching the plating. Set the unit on soft, clean surfaces.

(4) Use care and proper tools in removing and tightening screws to avoid burring the heads.

Packing

When you pack the unit for shipping:

(1) Wrap the unit in its original plastic bag to avoid marring the unit.
(2) Seal the carton with tape.

If you are returning the unit permanently (for credit), be sure to enclose:

The Manual(s)
The Registration/Warranty Card

Your dealer may charge you for any missing items.

If you are returning a unit for repair, do not enclose any of the above items.

Further advice on proper packing and shipping is included in the Manual (see Table of Contents).

Trouble

If you have problems with installation or operation:

(1) Check everything you have done so far against the instructions in the Manual. The information contained therein is based on our years of experience with OPTIMOD and broadcast stations.

(2) Check the other sections of the Manual (consult the Table of Contents) and search the text to see if there might be some suggestions regarding your problem.

(3) After reading the section on Factory Assistance, you may call Customer Service for advice during normal California business hours. The number is +1 909.860.6760.

IMPORTANT SAFETY INSTRUCTIONS

All the safety and operating instructions should be read before the appliance is operated.

Retain Instructions: The safety and operation instructions should be retained for future reference.

Heed Warnings: All warnings on the appliance and in the operating instructions should be adhered to.

Follow Instructions: All operation and user instructions should be followed.

Water and Moisture: The appliance should not be used near water (e.g., near a bathtub, washbowl, kitchen sink, laundry tub, in a wet basement, or near a swimming pool, etc.).

Ventilation: The appliance should be situated so that its location or position does not interfere with its proper ventilation. For example, the appliance should not be situated on a bed, sofa, rug, or similar surface that may block the ventilation openings; or, placed in a built-in installation, such as a bookcase or cabinet that may impede the flow of air through the ventilation openings.

Heat: The appliance should be situated away from heat sources such as radiators, heat registers, stoves, or other appliances (including amplifiers) that produce heat.

Power Sources: The appliance should be connected to a power supply only of the type described in the operating instructions or as marked on the appliance.

Grounding or Polarization: Precautions should be taken so that the grounding or polarization means of an appliance is not defeated.

Power-Cord Protection: Power-supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords at plugs, convenience receptacles, and the point where they exit from the appliance.

Cleaning: The appliance should be cleaned only as recommended by the manufacturer.

Non-Use Periods: The power cord of the appliance should be unplugged from the outlet when left unused for a long period of time.

Object and Liquid Entry: Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.

Damage Requiring Service: The appliance should be serviced by qualified service personnel when: The power supply cord or the plug has been damaged; or Objects have fallen, or liquid has been spilled into the appliance; or The appliance has been exposed to rain; or The appliance does not appear to operate normally or exhibits a marked change in performance; or The appliance has been dropped, or the enclosure damaged.

Servicing: The user should not attempt to service the appliance beyond that described in the operating instructions. All other servicing should be referred to qualified service personnel.

The Appliance should be used only with a cart or stand that is recommended by the manufacturer.

Safety Instructions (European)

Notice For U.K. Customers If Your Unit Is Equipped With A Power Cord.

WARNING: THIS APPLIANCE MUST BE EARTHED.

The cores in the mains lead are coloured in accordance with the following code:

GREEN and YELLOW - Earth BLUE - Neutral BROWN - Live

As colours of the cores in the mains lead of this appliance may not correspond with the coloured markings identifying the terminals in your plug, proceed as follows:

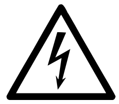
The core which is coloured green and yellow must be connected to the terminal in the plug marked with the letter E, or with the earth symbol, or coloured green, or green and yellow.

The core which is coloured blue must be connected to the terminal marked N or coloured black.

The core which is coloured brown must be connected to the terminal marked L or coloured red.

The power cord is terminated in a CEE7/7 plug (Continental Europe). The green/yellow wire is connected directly to the unit's chassis. If you need to change the plug and if you are qualified to do so, refer to the table below.

WARNING: If the ground is defeated, certain fault conditions in the unit or in the system to which it is connected can result in full line voltage between chassis and earth ground. Severe injury or death can then result if the chassis and earth ground are touched simultaneously.



Conductor		WIRE COLOR	
		Normal	Alt
L	LIVE	BROWN	BLACK
N	NEUTRAL	BLUE	WHITE
E	EARTH GND	GREEN-YELLOW	GREEN

AC Power Cord Color Coding

Safety Instructions (German)
Gerät nur an der am Leistungsschild vermerkten Spannung und Stromart betreiben.
Sicherungen nur durch solche, gleicher Stromstärke und gleichen Abschaltverhaltens ersetzen. Sicherungen nie überbrücken.
Jedwede Beschädigung des Netzkabels vermeiden. Netzkabel nicht knicken oder quetschen. Beim Abziehen des Netzkabels den Stecker und nicht das Kabel enfassen. Beschädigte Netzkabel sofort auswechseln.
Gerät und Netzkabel keinen übertriebenen mechanischen Beaspruchungen aussetzen.
Um Berührung gefährlicher elektrischer Spannungen zu vermeiden, darf das Gerät nicht geöffnet werden. Im Fall von Betriebsstörungen darf das Gerät nur Von befugten Servicestellen instandgesetzt werden. Im Gerät befinden sich keine, durch den Benutzer reparierbare Teile.
Zur Vermeidung von elektrischen Schlägen und Feuer ist das Gerät vor Nässe zu schützen. Eindringen von Feuchtigkeit und Flüssigkeiten in das Gerät vermeiden.
Bei Betriebsstörungen bzw. nach Eindringen von Flüssigkeiten oder anderen Gegenständen, das Gerät sofort vom Netz trennen und eine qualifizierte Servicestelle kontaktieren.

Safety Instructions (French)
On s'assurera toujours que la tension et la nature du courant utilisé correspondent bien à ceux indiqués sur la plaque de l'appareil.
N'utiliser que des fusibles de même intensité et du même principe de mise hors circuit que les fusibles d'origine. Ne jamais shunter les fusibles.
Eviter tout ce qui risque d'endommager le câble seceur. On ne devra ni le plier, ni l'aplatir. Lorsqu'on débranche l'appareil, tirer la fiche et non le câble. Si un câble est endommagé, le remplacer immédiatement.
Ne jamais exposer l'appareil ou le câble ä une contrainte mécanique excessive.
Pour éviter tout contact averc une tension électrique dangereuse, on n'oouvrira jamais l'appareil. En cas de dysfonctionnement, l'appareil ne peut être réparé que dans un atelier autorisé. Aucun élément de cet appareil ne peut être réparé par l'utilisateur.
Pour éviter les risques de décharge électrique et d'incendie, protéger l'appareil de l'humidité. Eviter toute pénétration d'humidité ou fr liquide dans l'appareil.
En cas de dysfonctionnement ou si un liquide ou tout autre objet a pénétré dans l'appareil couper aussitôt l'appareil de son alimentation et s'adresser à un point de service aprésvente autorisé.

Safety Instructions (Spanish)
Hacer funcionar el aparato sólo con la tensión y clase de corriente señaladas en la placa indicadora de características.
Reemplazar los fusibles sólo por otros de la misma intensidad de corriente y sistema de desconexión. No poner nunca los fusibles en puente.
Proteger el cable de alimentación contra toda clase de daños. No doblar o apretar el cable. Al desenchufar, asir el enchufe y no el cable. Sustituir inmediatamente cables dañados.
No someter el aparato y el cable de alimentación a esfuerzo mecánico excesivo.
Para evitar el contacto con tensiones eléctricas peligrosas, el aparato no debe abrirse. En caso de producirse fallos de funcionamiento, debe ser reparado sólo por talleres de servicio autorizados. En el aparato no se encuentra ninguna pieza que pudiera ser reparada por el usuario.
Para evitar descargas eléctricas e incendios, el aparato debe protegerse contra la humedad, impidiendo que penetren ésta o líquidos en el mismo.
En caso de producirse fallas de funcionamiento como consecuencia de la penetración de líquidos u otros objetos en el aparato, hay que desconectarlo inmediatamente de la red y ponerse en contacto con un taller de servicio autorizado.

Safety Instructions (Italian)
Far funzionare l'apparecchio solo con la tensione e il tipo di corrente indicati sulla targa riportante i dati sulle prestazioni.
Sostituire i dispositivi di protezione (valvole, fusibili ecc.) solo con dispositivi aventi lo stesso amperaggio e lo stesso comportamento di interruzione. Non cavallottare mai i dispositivi di protezione.
Evitare qualsiasi danno al cavo di collegamento alla rete. Non piegare o schiacciare il cavo. Per staccare il cavo, tirare la presa e mai il cavo. Sostituire subito i cavi danneggiati.
Non esporre l'apparecchio e il cavo ad esagerate sollecitazioni meccaniche.
Per evitare il contatto con le tensioni elettriche pericolose, l'apparecchio non deve venir aperto. In caso di anomalie di funzionamento l'apparecchio deve venir riparato solo da centri di servizio autorizzati. Nell'apparecchio non si trovano parti che possano essere riparate dall'utente.
Per evitare scosse elettriche o incendi, l'apparecchio va protetto dall'umidità. Evitare che umidità o liquidi entrino nell'apparecchio.
In caso di anomalie di funzionamento rispettivamente dopo la penetrazione di liquidi o oggetti nell'apparecchio, staccare immediatamente l'apparecchio dalla rete e contattare un centro di servizio qualificato.

WARNING

This equipment generates, uses, and can radiate radio-frequency energy. If it is not installed and used as directed by this manual, it may cause interference to radio communication. This equipment complies with the limits for a Class A computing device, as specified by FCC Rules, Part 15, subject J, which are designed to provide reasonable protection against such interference when this type of equipment is operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference. If it does, the user will be required to eliminate the interference at the user's expense.

WARNING

This digital apparatus does not exceed the Class A limits for radio noise emissions from digital apparatus set out in the radio Interference Regulations of the Canadian Department of Communications. (Le present appareil numerique n'emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques [de las class A] prescrites dans le Reglement sur le brouillage radioelectrique edicte par le ministere des Communications du Canada.)

IMPORTANT

Perform the installation under static control conditions. Simply walking across a rug can generate a static charge of 20,000 volts. This is the spark or shock you may have felt when touching a doorknob or some other conductive surface. A much smaller static discharge is likely to destroy one or more of the CMOS semiconductors employed in OPTIMOD. Static damage will not be covered under warranty.

There are many common sources of static. Most involve some type of friction between two dissimilar materials. Some examples are combing your hair, sliding across a seat cover or rolling a cart across the floor. Since the threshold of human perception for a static discharge is 3000 volts, you will not even notice many damaging discharges.

Basic damage prevention consists of minimizing generation, discharging any accumulated static charge on your body or workstation, and preventing that discharge from being sent to or through an electronic component. You should use a static grounding strap (grounded through a protective resistor) and a static safe workbench with a conductive surface. This will prevent any buildup or damaging static.

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Thanks go to Greg Ogonowski of Modulation Index LLC for many of the tutorial diagrams in this manual.

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AUDIO PROCESSING

This product exclusively distributed and supported by:

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Operating Manual

OPTIMOD-PCn 1600/1601/1602

Digital Audio Processor Software
for Intel/Windows PCs

Manual for Software Version 0.9.17

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Table of Contents

Section 1

Introduction

CAUTION: To enable the 1600 to exploit the full power of its Intel CPU while ensuring smooth, uninterrupted operation, hardware products incorporating OPTIMOD-PCn ship from the factory with a specific Windows operating system and software configuration. **DO NOT RECONFIGURE THE OPERATING SYSTEM AND/OR INSTALL ADDITIONAL SOFTWARE UNLESS INSTRUCTED TO DO SO BY THE VENDOR OF YOUR SYSTEM OR BY SPECIFIC INSTRUCTIONS IN THIS MANUAL.** Doing so may cause the system to operate improperly and will make Technical Support unavailable to you because we cannot support non-tested OS configurations and third-party software.

If you have purchased the software-only version of 1600 PC, **we strongly recommend running it on a clean Windows installation and using our recommended sound I/O hardware**. Other software can compete for CPU cycles, causing audio glitches and interruptions. We can recommend software (like playout systems) that is known to play well with 1600 PC. Other hardware may have driver problems that prevent glitch-free operation.

About this Manual

The Adobe pdf form of this manual contains numerous hyperlinks and bookmarks. A reference to a numbered step or a page number (except in the Index) is a live hyperlink; click on it to go immediately to that reference.

To help you find the information you need, this manual has an index (starting on page 7-1) and a Table of Contents. To search for a specific word or phrase, you can also use the Adobe Acrobat Reader's text search function.

- Section 1 contains general information about OPTIMOD-PCn and describes how to use it in your specific application. The most important material for typical users appears first. OPTIMOD-PCn uses the contemporary concept of "target loudness" (per ITU-R BS.1770), so the output level control works differently than you might expect (it adjusts output peak headroom but not loudness), and the explanation of target loudness (Setting Loudness starting on page 1-20) is important to understand.

- Section 2 explains how to install and set up OPTIMOD-PCn.

- Section 3 explains how to customize OPTIMOD-PCn's sound to your requirements and contains descriptions of all tuning controls for the audio processing.

- Section 4 describes the software that implements OPTIMOD-PCn's functionality.
- Section 5 documents OPTIMOD-PCn's control API.
- Section 6 contains OPTIMOD-PCn's specifications.
- Section 7 is the Index.

The OPTIMOD-PCn 1600 Digital Audio Processor

For a list of features, see *Summary of OPTIMOD-PCn's Features* starting on page 1-7.

The Model 1600 OPTIMOD-PCn is audio processing software that is available both as a software-only product for approved Windows® computers and pre-installed on a computer that can be configured at the factory to run advanced audio processing software (called OPTIMOD-PCn in this manual) and MPEG-4 AAC/HE-AACv2 streaming audio codecs supplied by Modulation Index (StreamS Live Encoder®[1]) software natively on its Intel processor. Depending on its ordered configuration, the 1600 can run multiples instances of monophonic, stereo, 5.1, or 7.1 processing, and these can be mixed and matched as required. The 1600 is equally suited for mastering, netcasts and digital radio broadcasting.

> While OPTIMOD-PCn can be used for video applications, video applications in SDI or video-over-IP facilities require de-embedding and re-embedding audio. In all applications, you must apply a video delay that matches the delay of the audio processed through OPTIMOD-PCn, which may be a much as 1 second. OPTIMOD-PCn ships with a full complement of sound-for-picture presets, and this manual includes instructions for using the OPTIMOD-PCn in video applications.

From the ground up, OPTIMOD-PCn 1600 was designed for professionals. It offers broadcast-quality digital signal processing that is suitable for both live streaming and on-demand programming. OPTIMOD-PCn uses the power of Intel's x86 architecture provide a consistent, well-produced sound to the consumer by performing phase skew correction, stereo enhancement, automatic gain control (AGC), equalization, multiband gain control, stereo-to-surround upmixing, peak-level control, and automatic loudness control.

OPTIMOD-PCn's setup, metering, and subjective loudness control incorporate contemporary concepts of "target loudness" based on the ITU-R BS.1770 loudness measurement algorithm (see *About Target Loudness and ITU-R BS.1770* starting on page 1-16) and on our third-generation refinements to the CBS (Jones & Torick) loudness controller and loudness metering technology. (See *Level and Subjective Loudness Metering in OPTIMOD-PCn* starting on page 1-25.)

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[1] https://www.indexcom.com/products/encoder/

OPTIMOD-PCn supports audio-over-IP via any standard that provides a Windows sound device driver, such as RAVENNA and Dante. An optional sound card provides digital AES3 inputs and outputs that are compatible with industrial and broadcast equipment, and that work with consumer electronics too. OPTIMOD-PCn also includes mixers and software Wave I/O that interfaces directly with audio encoders running on the same computer as OPTIMOD-PCn.

OPTIMOD-PCn was designed to deliver a high-quality sound while simultaneously increasing the average level on the channel substantially beyond that achievable by "recording studio"-style compressors and limiters. Because such processing can exaggerate flaws in the source material, it is very important that the **source audio be as clean as possible.**

For best results, **feed OPTIMOD-PCn unprocessed audio.** No other audio processing is necessary or desirable.

In digital radio applications, if you wish to place level protection prior to your studio/transmitter link (STL), we suggest using another OPTIMOD-PCn, running an "AGC" preset. Other types of AGC systems may adversely affect the audio.

Hardware and Software Requirements

The OPTIMOD-PCn system consists of a core audio processing engine and a graphical interface application (called 1600PC) to control the audio processing engine. The core audio processing engine is a Windows Service (called OptimodService3) that can spawn up to 16 independent audio processors, depending on the number of processors you have purchased.

1600PC is an application that can run on the same computer as the OPTIMOD-PCn Service or on any computer connected to the same network. Only the Service is copy-protected; your software license allows you to install as many copies of 1600PC on as many computers as you wish. Only one instance of 1600PC can be connected to a given audio processor at any one time, although when a given Service spawns more than one audio processor, each such processor may be connected to its own 1600PC instance.

OPTIMOD-PCn requires a substantial amount of computing power. To achieve most reliable, glitch-free operation, it must be used with carefully selected and configured hardware.

Operating System

Windows 7 / 8.1 / 10, Professional edition or higher.

- Do not run general-purpose programs (like Microsoft Office) on the same computer.
- Turn off Windows Defender or other anti-malware software.

> Do not use the computer running the OPTIMOD-PCn Service for web browsing or email.

- You may run approved audio playout systems (like Modulation Index's RadioDJ) and approved streaming encoder software (like StreamS Live Encoder).

- Within the limits described below, you may interconnect the playout software, OPTIMOD-PCn, and the encoder via Virtual Audio Cable[2]. To achieve highest audio quality, it is important to configure VAC as specifically as possible in terms of sample rate, number of audio channels, and word length.

  To avoid glitches or delay build-up, the input and output sample rate of a given application must be precisely locked together, but the VAC clock frequency cannot be locked to a hardware sound device. Moreover, older versions of VAC did not support locking multiple VAC "cables" to the same (internal) clock. *Hence VAC can only be used safely with a given application if that application uses VAC for both input and output and you are using the current version of VAC*.

- In Windows CONTROL PANEL > ALL CONTROL PANEL ITEMS > PERFORMANCE INFORMATION AND TOOLS > PERFORMANCE OPTIONS, choose ADJUST FOR BEST PERFORMANCE.

- Turn off Windows Automatic Updates, as these can force the computer to reboot and interrupt audio.

- Turn off Windows Sounds.

- In Windows 8 and higher, turn off Fast Startup.

CPU

OPTIMOD-PCn requires a genuine Intel Core2[3] (3 GHz or faster using Intel's 45 nm manufacturing process), i3, i5, or i7, CPU of any generation. OPTIMOD-PCn uses Intel vector floating point instructions to achieve the required computational efficiency. The baseline OPTIMOD-PCn code requires the SSE4.1 vector instruction set. Core 2 and all i-series processors support SSE4.1. You may also use an Intel Xeon processor if it supports SSE4.1.

> Compared to the i-series, Intel processors like the Pentium and Celeron series, even if they support SSE4.1, have substantially lower computing power for a given clock speed and are insufficiently powerful to support all of OPTIMOD-PCn's features.

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[2] http://software.muzychenko.net/eng/vac.htm

[3] Older model Core 2 processors using Intel's 65 nm manufacturing process do not support SSE4.1 and cannot be used with OPTIMOD-PCn. Before installing OPTIMOD-PCn on a PC with a Core 2 CPU, do an Internet search to see if its specific model of processor supports SSE4.1.

OPTIMOD-PCn is not optimized for AMD processors, and we have not tested it on these processors.

We recommend sixth-generation i7 CPUs or higher. These are approximately 1.7 times as efficient than the first generation because they have a more efficient architecture and support the Intel AVX2 vector instruction set, which OPTIMOD-PCn exploits if available. When you run the OPTIMOD-PCn installer, it automatically detects the vector instruction set that the computer's CPU supports and installs the correct software. Attempting to run OPTIMOD-PCn for AVX2 on a CPU that does not support it will cause OPTIMOD-PCn to crash.

Although the operating system may quickly switch a given audio Processor between CPU cores, each Processor spawned by the Service cannot exploit more CPU resources than are available for a single core, which may be a Hyperthreaded core. On the other hand, several Processors can share one core if that core has enough resources to run them all smoothly.

In addition to the number of audio processors specified in your purchased license, a further limitation is the computational capacity of the i-series CPU, which depends mainly on its clock speed. Different configurations of an audio processor use different amounts of CPU. The main determining factors are whether the output is upmixed to 5.1 surround, whether the MX peak limiter is activated, and whether the MX limiter (if activated) uses SOFT or HARD overshoot compensation.

HARD is only beneficial when you are trying to achieve BS.1770 loudness levels higher than −6 LUFS. This is rarely desired and usually a bad idea anyway because loudness this high compromises audio quality.

Table 1-1 on page 1-5 shows the theoretical minimum CPU clock speed for various

Processing Configuration	**Per-core Clock (GHz) for 100% utilization**	
[Note: requirements are approximate and can vary with system configuration.]	1st-gen i7 (SSE4.1)	6th-gen i7 (AVX2)
5.1 surround + one downmix (no MX)	1.7 GHz	1.3 GHz
5.1 surround + one downmix (soft MX)	3.5 GHz	2.2 GHz
5.1 surround (no MX)	1.7 GHz	1.2 GHz
5.1 surround (soft MX)	3.5 GHz	2.2 GHz
Stereo + upmix (no MX)	1.8 GHz	1.1 GHz
Stereo + upmix (soft MX)	2.8 GHz	1.8 GHz
Stereo + upmix (hard MX)	3.6 GHz	2.5 GHz
stereo (no MX)	1.5 GHz	0.8 GHz
stereo (soft MX)	2.3 GHz	1.3 GHz
stereo (hard MX)	2.6 GHz	1.6 GHz
mono (no MX)	0.6 GHz	0.3 GHz
mono (soft MX)	1.5 GHz	0.7 GHz
mono (hard MX)	1.7 GHz	0.8 GHz

Table 1-1: CPU hardware requirements for one Intel i-series processor hardware core. Requirements intermediate-generation i-series can be interpolated.

combinations of features in 1^{st}-generation and 6^{th}-generation i-series processors. In practice, it is wise to allow at least 20% headroom above these speeds to accommodate other demands that the operating system makes on the CPU.

To calculate the number of instances that you can run per core, calculate the ratio of the “clock” shown in Table 1-1 and the clock speed of your processor. If it is less than 0.5, you can run two instances; if it is less than 0.33, you can run three instances. For example, a 2.6 GHz 6^{th}-generation processor could theoretically run two instances of stereo in each core with soft MX limiting (2 x 1.3 GHz = 2.6 GHz). A good choice would be a 3.3 GHz processor, which provides a comfortable amount of headroom. A four-core 6^{th}-generation 3.6 GHz i7 could run 8 instances of stereo/soft MX or 12 instances of stereo/no MX with enough processing power left over to run the operating system, playout systems, and streaming encoders.

If you use Hyperthreading (which we recommend), be aware that one physical CPU core divides its available resources between two Hyperthreaded cores, and Table 1-1 indicates the physical resources that are required.

> Intel claims that two Hyperthreaded cores can provide as much as 1.3 times the processing power available from their host physical core in non-Hyperthreaded mode. As this is application- and OS-dependent, you will have to experiment to see how much (if any) performance bonus you get from Hyperthreading in your hardware.

When you are running four or more processors, the CPU requirements are slightly higher (by about 8%) to compensate for the fact that the CPU’s memory cache is shared among the various processors.

When heavily loaded by multiple OPTIMOD-PCn Processors, the CPU will generate a significant amount of heat. This is particularly true of first and second-generation Intel i7 CPUs, which may dissipate as much as 130 watts. Be sure that the CPU has adequate heat sinking, the chassis has adequate forced-air cooling, and the computer is located in a temperature-controlled environment. Insufficient cooling can shorten the life of the CPU and can cause its clock speed to be throttled back automatically and unexpectedly, limiting the number of Processors than it can run without glitching. When first setting up OPTIMOD-PCn, It is wise to verify that your computer has adequate cooling by running a utility that can indicate CPU core temperature.

Audio I/O Devices

For multichannel I/O, we recommend and use Lynx® sound cards[4] because these cards tend to have stable drivers and are designed for professional users.

If you need low-latency headphone monitoring and mono or stereo I/O, we recommend the Orban Optimod-PC 1101e card[5], which has onboard, low-latency audio

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[4] https://www.lynxstudio.com/

[5] https://www.orban.com/optimodpc-1101e/

processing. While this processing is not as capable or feature-rich as Optimod-PCn, it is a good choice to drive talent headphones. Functionally, Optimod-PC looks like two sounds cards to Windows and has versatile mixing and audio routing capabilities.

OPTIMOD-PCn supports audio-over-IP via any standard that provides a Windows sound device driver, such as RAVENNA and Dante.

Summary of OPTIMOD-PCn's Features

The list of features below pertains only to the OPTIMOD-PCn audio processor and does not include the features of StreamS Live software, which may also be installed in the 1600 hardware. The features shown are for the full 1600PCn processor. Models 1601PCn and 1602PCn (at lower cost) remove some features. See *1602PCn Features* on page 1-12 and *1601PCn Features* on page 1-12.

General Features

- **Optimod-quality digital audio processing,** running natively on an Intel x86 computer, pre-processes audio in real time for consistency and loudness before it is transmitted or recorded.

- Available in **mono/stereo and surround** versions (priced separately). **One server computer supports up to 16 audio processors**, which may be any combination of mono/stereo and surround processors.

- **Applications** include netcasting, DAB and other dedicated digital radio services, FMExtra™ and other digital subcarriers, sound-for-picture, mastering, audio production, and many others.

- Can be used as a **pre-processor for existing hardware Optimod processors,** adding features like **parallel compression** and the "**Multipath Mitigator**" phase corrector.

- Basic **offline file processing** functionality allows you to apply faster than real-time Optimod PCn processing to single or multiple wav files.

- Incorporates modern **"target loudness" concepts** (including those specified in **EBU R128** and **ATSC A/85**) **using the ITU-R BS.1770 loudness model.** Allows you to easily **set and verify the target loudness of the output**.

- The audio processor runs as a **Windows Service** on its host computer, so it will **start automatically with Windows** and **run reliably in the background**.

- A responsive, smooth, easy-to-use **graphic control application** runs on local or remote PCs and can control any number of OPTIMOD-PCn audio Processors, either locally (via a localhost TCP/IP connection) or in other 1600s on your network via **TCP/IP** addressing.

The Control application allows you complete flexibility to create **your own custom presets**, to save as many as you want to your local hard drive, and to recall them at will.

- By use of third party Windows Audio drivers, is compatible with **audio-over-IP input/output** (such as Ravenna®, LiveWire®, Dante®, and WheatNet®).

- Compatible with **all Windows sound devices** with stable drivers supporting the Windows **WASAPI** standard. (WASAPI stands for Windows Audio Session API and was first introduced in Windows Vista.)

- **Precisely controls peak levels** to prevent overmodulation or codec overload. While **primarily oriented toward "flat" media**, OPTIMOD-PCn can also provide **preemphasis limiting** for the two standard preemphasis curves of 50μs and 75μs. This allows it to protect preemphasized **satellite uplinks and similar channels** where protection limiting or light processing is required. It can also be used to process analog television FM aural carriers in television applications because these are usually processed lightly compared to FM radio.

- **PreCode**™ technology manipulates several aspects of the audio to minimize artifacts caused by low bitrate codecs, ensuring consistent loudness and texture from one source to the next. PreCode includes special audio band detection algorithms that are energy and spectrum aware. This can improve codec performance on some codecs by reducing audio processing induced codec artifacts, even with program material that has been preprocessed by other processing than Optimod.

- **Controls audio bandwidth** as necessary to accommodate the transmitted sampling frequency, obviating the need for extra, overshooting anti-aliasing filters in downstream equipment. OPTIMOD-PCn's high frequency bandwidth can be switched instantly (typically in 1 kHz increments) between 10.0 kHz and 20 kHz. 20 kHz is used for highest-quality systems. 15 kHz codec bandwidth may **help low bitrate lossy codecs sound better than they do when fed full 20 kHz bandwidth audio**. 15 kHz is well matched to the codec used the iBiquity® HD-AM system.

- Intended for program equalization and mastering, a **sweepable gentle-slope lowpass filter (6 to 24 dB / octave)** is available in addition to the sharp-cutoff lowpass filter.

- For specialized purposes like speech processing, a **sweepable highpass filter** with four selectable slopes (6 to 24 dB / octave) is available. **Music and speech modes have separate cutoff frequencies and slopes**, so the **automatic speech/music detector can change filter characteristics**.

- The highpass and lowpass filters provide **click-free switching of their cutoff frequencies,** so they these may be changed in the middle of program material.

- A **DC removal filter** with a 0.1 Hz −3 dB low frequency cutoff removes DC offset from source material **without introducing overshoot and tilt into low-frequency waveforms**.

- OPTIMOD-PCn ships with **many standard presets**, designed to accommodate almost any programming format. There are also special-purpose, no-compromise presets for **mastering** and **pure peak limiting**.

- A **Bypass Test Mode** facilitates broadcast system **test and alignment** or "proof of performance" tests.

- Audio processing can be **smoothly activated and defeated on-air via a delay-matched pass-through mode**, allowing programs that can benefit from full dynamic range to pass through OPTIMOD-PCn without dynamics compression except **for safety limiting using MX or non-MX limiter modes**.

- OPTIMOD-PCn contains a built-in **line-up tone generator**, facilitating quick and accurate level setting in any system. **Low-distortion sinewave, non-overshooting squarewave, and high-precision pink noise** test signals are available.

- **Dongle-based copy protection** allows the software to be easily moved from one computer to another **without elaborate re-authorization procedures**. **No Internet connection is required**. The **graphic control application is not copy-protected** and can be installed on an unlimited number of computers.

Audio Processing Features

- OPTIMOD-PCn **can increase the density and loudness** of the program material by multiband compression and sophisticated peak limiting, improving the consistency of the station's sound and increasing loudness and definition remarkably, without producing unpleasant side effects.

- **Automatic left/right phase-skew corrector** can **eliminate comb-filtering artifacts** in a mono downmix. When used as a preprocessor for Optimod-FM, it can **reduce multipath distortion** by minimizing energy in the stereo subchannel, which carries the L–R signal.

- **AGC** can operate in **left/right** or **sum/difference** modes.

- **Bass can be made monophonic,** with crossover frequencies of 80 or 100 Hz.

- Two-Band **automatic gain control** with a **phase-linear crossover**, adjustable **band coupling**, and **window gating** compensates for widely varying input levels. The AGC **rides gain** over an adjustable range of up to 25dB, compressing dynamic range and compensating for operator gain-riding errors and for gain inconsistencies in automated systems

- **Five-Band** compression with **selectable phase-linear and allpass crossover topologies** provides a consistent, "processed" sound, free from undesirable side effects.

  Band-coupling controls allow the gain differences between adjacent bands of the five-band compressor to be constrained to any desired value, allowing you to **preserve as much of the frequency balance of the original program material as desired** unless doing so would otherwise cause objectionable spectral gain intermodulation artifacts. Combined with the phase-linear crossover, this functionality can be used in lieu of the two-band compression found in earlier Optimods.

  Parametric soft-knee compression curves with adjustable ratio allow you to **fine-tune the audio to your exact requirements** and make OPTIMOD-PCn an excellent **mastering processor**.

  The five-band compressor can be operated in both **inline** and **parallel** modes. Parallel mode drives downstream processing with the sum of the input and output of the multiband compressor. This **increases the loudness of quiet material while preserving the dynamics and punch of loud material**. Typical applications include preparing **classical music for in-flight entertainment or other noisy environments,** or as preprocessor for Optimod-FM.

- **Stereo Enhancer** is based on Orban's patented analog 222 Stereo Enhancer, which increases the energy in the stereo difference signal (L−R) whenever a transient is detected in the stereo sum signal (L+R). Gating circuitry **prevents over-enhancement** and undesired enhancement on slightly unbalanced mono material. **Complements the Optimix upmixer**, increasing the sense of envelopment and space while maintaining **excellent downmix compatibility**.

- Shelving **bass equalizer** and **four-band parametric equalizer** let you color the audio to your exact requirements. To facilitate A/B comparisons, **equalizers can be bypassed** individually and globally.

- **Dynamic High-Frequency Enhancer** constantly monitors the ratio of HF to broadband energy in the incoming audio and can **automatically re-equalize it** to achieve a target balance between broadband and HF energy.

- **Subharmonic Synthesizer** generates punchy bass from bass-shy material. It is particularly useful for older music recordings.

- Orban's exclusive **MX peak limiter technology** uses a **psychoacoustic model** to achieve an **unprecedentedly favorable tradeoff between loudness, transient punch, and distortion artifacts**.

  For applications where target loudness is below approximately −12 LkFS/LUFS, you can reduce CPU load by approximately 50% without compromising audio quality (compared to full MX limiter operation) by defeating the MX limiter and

instead using our smooth, **low-IM look-ahead limiter** for the very light peak limiting required at these low loudness levels. This is the normal mode of operation for **sound-for-picture** applications, where target loudness is typically −23 LUFS or −24 LUFS and peak limiting rarely occurs at all.
In all modes of operation, the peak limiter offers **"true peak" control** by oversampling the peak limiter's sidechain at 192 kHz or higher. This allows OPTIMOD-PCn **to prevent clipping in a playback device's analog signal path** by predicting and controlling the analog peak level following the playback device's reconstruction filter.

Without true peak control, analog clipping can occur even if all peak values of the digital samples are below 0 dBFS. This phenomenon has also been termed "**0 dBFS+**."

Thanks to true peak control, **sample rate conversion**, unless it removes high frequency program energy or introduces group delay distortion, **cannot cause sample peaks to increase more than 0.5 dB**. For example, sample rate conversion from 48 kHz to 44.1 kHz is highly unlikely to cause sample peak clipping in the 44.1 kHz audio data.

Accuracy is typically 0.15 dB even with heavy peak limiting, so the output level can be set to −0.2 dBFS without true peak levels exceeding 0 dBFS.

- Second-generation **Optimix**™ stereo → 5.0 surround **upmixer** provides **uncolored automatic upmixing**, plus phase correction that ensures that the **center channel is always crisp and intelligible**. 5.1-compatible. Provides **excellent downmix compatibility**. Can **wrap the virtual soundstage around the listener** (put the listener "in the middle of the band") or **place the stage primarily in the front** (put the listener "in the audience").

- **Orban Stereo Synthesizer** with excellent **downmix compatibility. "Wide" and "narrow" modes** (based on Orban's classic analog 275A Stereo Synthesizer) can create an attractively spacious stereo or surround output from mono program material. Synthesis can be invoked manually, or activated automatically by sensing silence in the right channel input.

- OPTIMOD-PCn includes third-generation **CBS Loudness Controllers**™ for DTV applications. **Separate loudness controllers are available in both the surround and downmix processing chains**. The third-generation improvements reduce annoyance more than simple loudness control alone, doing so **without audible gain pumping**. Attack time is fast enough to prevent audible loudness overshoots, so the control is **smooth and unobtrusive**. Material processed by the CBS Loudness Controller has been shown to be well controlled when measured with a long-term loudness meter using the BS.1770 standard. (See *Appendix A: Using the ITU BS.1770 and CBS Loudness Meters to Measure Loudness* Controller Performance starting on page 3-93.)

- **BS.1770-4** and **CBS Loudness Meters™** measure the subjective loudness of the 1600's output and are displayed in the 1600's control application meter window. When downmix processing is active, there are **two independent loudness controllers** and **two loudness meters** available.

- A **BS.1770 Safety Limiter** follows the CBS Loudness Controller, When activated, it can further improve the measured performance using the BS.1770 meter and which was added for the benefit of organizations with strict objective limits on the indication of a BS.1770-2 meter regardless of the actual subjective loudness as determined by human listeners.

- A pure peak limiting preset is available. It allows OPTIMOD-PCn to perform **very high quality peak limiting in mastering applications**.

- OPTIMOD-PCn can be used as a **studio AGC** (including peak limiting) to protect a studio-to-transmitter link (STL), optimally using the STL's native dynamic range.

- Uses **"multirate" digital signal processing**. Internal processing always occurs at sample rates from **48 kHz** to **256 kHz** as needed, and provides **20 kHz** audio bandwidth (unless specifically constrained by the user-adjustable lowpass filter). Built-in **high-quality synchronous sample rate converters** facilitate interfacing with **44.1 kHz, 96 kHz and 192 kHz** systems.

- Uses **32-bit or 64-bit floating-point arithmetic** as appropriate. Can interface to **16-bit and 24-bit** Windows audio devices.

1601PCn Features

1601PCn does not provide the following features from 1600PCn:

- File encoder
- Optimix/Stereo Synthesizer
- MX Limiter
- Parallel Compression Mode
- Subharmonic Synthesizer
- CBS Loudness Meter (leaving the BS.1770 meter in place)
- Loudness controller

1602PCn Features

1602PCn does not provide the following features from 1600PCn:

- File encoder

- Optimix/Stereo Synthesizer
- MX Limiter
- Parallel Compression Mode
- Subharmonic Synthesizer
- CBS Loudness Meter (leaving the BS.1770 meter in place)
- Loudness controller
- High-Frequency Enhancer
- Phase Corrector
- Stereo Enhancer
- Mono Bass
- Crossover Mode (is always ALLPASS)
- Dual-Mono AGC

Why use OPTIMOD-PCn?

Audio Processing: Making Broadcasts/Netcasts Sound Professional

Professional broadcasters would never consider going on the air without audio signal processing. They consider it a vital aspect of their program content. This carefully crafted content is what holds listeners and keeps them coming back. Since 1975, Optimod algorithms have dominated the world market for professional radio and television audio processing and have been improved continuously since then.

To make scenes flow naturally through cuts, all broadcast television and Hollywood movies have every scene color-corrected by a colorist, who adjusts hue, saturation, gamma, and gray scale to achieve the "look" that the director intends, whether natural or stylized. OPTIMOD-PCn processing is like "audio color correction" for your programming. It subtly and automatically modifies the loudness, spectral balance, and texture of program elements to ensure that they flow smoothly into one another. OPTIMOD-PCn's various factory presets allow you to choose the sonic "look" of your programming while ensuring that this texture is maintained throughout your programming.

OPTIMOD-PCn audio processing is appropriate for all digital transmission media and channels. It tailors your audio signal to help you compete in audio netcasting, HD Radio® (both primary and multicast digital channels), DAB, DRM, and other dedicated digital radio services, FMExtra™ and other digital subcarriers, mastering, audio production, and many others.

Video-oriented presets use OPTIMOD-PCn's built-in CBS Loudness Controller™ to make OPTIMOD-PCn well suited for sound-for-picture applications, including HDTV, DVB-x digital television, and audio/video netcasting. OPTIMOD-PCn also includes a defeatable "BS.1770 Safety Limiter" that follows the CBS Loudness Controller in the signal path. The BS.1770 Safety Limiter can be set to prevent the indication of a BS.1770 long-term meter with ten-second or longer integration time from being higher than a preset threshold value: anywhere from 0 dB to +6 dB, as set by the BS.1770 THRESHOLD control. For specific instructions on using OPTIMOD-PCn in sound-for-picture applications, refer to *Sound for Picture Applications: Controlling Dynamic Range* on page 2-8 and *Sound for Picture Presets* on page 3-42.

While primarily oriented toward "flat" media, OPTIMOD-PCn can also provide preemphasis limiting for the two standard preemphasis curves of 50 µs and 75 µs. It provides a "light" version of Orban's exclusive MX peak limiting technology (first developed for Optimod-FM 8600) to facilitate processing for preemphasized channels. This allows OPTIMOD-PCn to protect preemphasized analog satellite uplinks and similar channels where protection limiting or light processing is required, and makes it ideal for use as a studio AGC driving transmitter-located Optimods. It also allows OPTIMOD-PCn to process audio for analog television transmission. We pro-

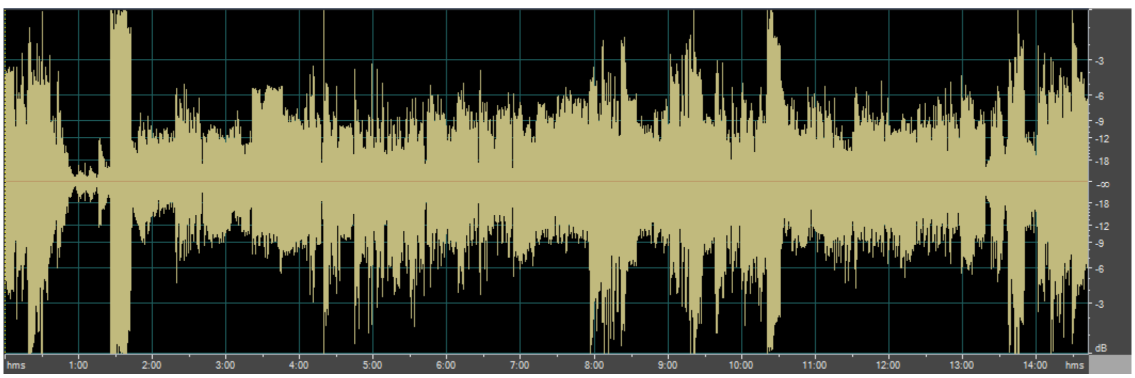

Figure 1-1: Unprocessed Audio

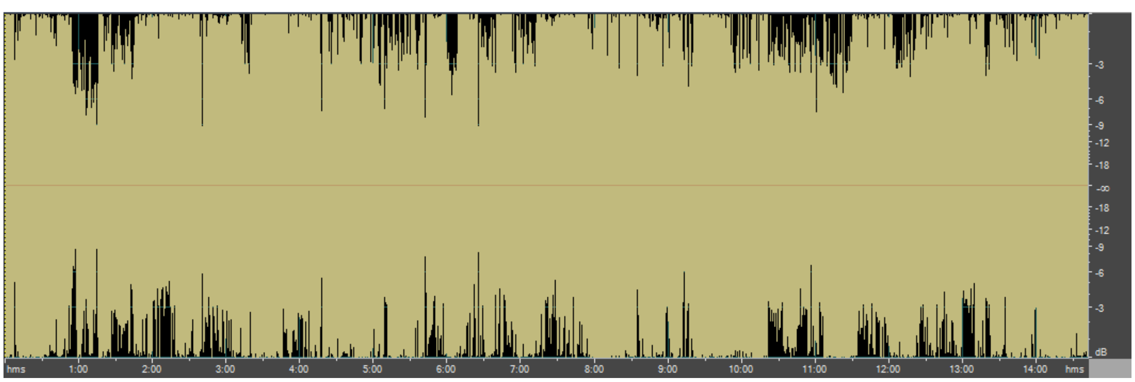

Figure 1-2: Same Audio Processed Through OPTIMOD-PCn

vide a number of factory presets for this purpose.

There are many gain/peak control devices and software available to perform dynamics processing. Many of these tools are designed for recording studio applications as effects compressors/limiters for individual microphone or instrument tracks. These devices' controls need to be tuned carefully for the specific material being processed—they are not "set and forget" processors. Moreover, most do not process mixed program material without introducing objectionable audible artifacts, particularly when called upon to gain-ride input material having widely varying levels. The details and implementations are all-important.

A broadcast audio processor should ideally be "seen but not heard." Optimod processing algorithms simultaneously control audio gain, loudness, and peaks, artistically, musically, and naturally, to give the illusion that processing is not taking place. Moreover, Optimod algorithms intelligently adapt themselves to the input program material. Once OPTIMOD-PCn is tuned for the sound texture required for the broadcast or netcast format (which is made easy by OPTIMOD-PCn's many format-specific presets), it will provide excellent consistency regardless of the level or texture of the original program material. OPTIMOD-PCn's automatic gain control and equalization achieve a consistent sound, while accurate peak control maximizes loudness. Booming bass is tightened; weak, thin bass is brought up; highs are always present and consistent in level.

OPTIMOD-PCn is also an excellent mastering processor, offering soft knee multiband compression with knee and ratio controls available separately for each band. This is followed by a peak limiter using Orban's MX technology, which uses a psychoacoustic model and can typically achieve more than 12 dB of gain reduction before it produces objectionable artifacts—this limiter is exceptionally loud, clean, and punchy. MX limiting is effective with both "flat" and pre-emphasized transmission channels. See *Production and Mastering* on ape 3-86.

An API provides complete remote administration over TCP/IP. The OPTIMOD-PCn Service application hosts a TCP/IP terminal server to allow external control of the OPTIMOD-PCn Service from either a Telnet/SSH client or a custom third party application. See *OPTIMOD 1600PCn Control API* starting on page 5-1.

OPTIMOD-PCn comes with many great-sounding presets that make it easy to create a sonic texture that's just right for your target audience. If you want to customize a preset, you can start with an easy LESS-MORE control, and, if that's not enough, tweak dozens of parameters to hone your sound to perfection. OPTIMOD-PCn's deep interface will never hold you back as your processing expertise increases, yet its carefully crafted design insulates you from the details if you need great sound right now. See *Table 3-1: Protection, AGC and Mastering Presets* on page 3-34, *Table 3-2: Radio-Style Presets* on page 3-38, and *Table 3-3: Sound-For-Picture Presets* on page 3-42.

Without OPTIMOD-PCn processing, audio can sound dull, thin, or inconsistent in any combination. OPTIMOD-PCn's multiband processing automatically levels and re-equalizes its input to the "major-market" standards expected by the mass audience.

Broadcasters have known for decades that this polished, “produced” sound attracts and holds listeners.

You can expect a considerable increase in loudness from OPTIMOD-PCn processing compared to unprocessed audio (except for audio from recently mastered CDs, which are often over-processed in mastering). Broadcasters generally believe that loudness relative to other stations attracts an audience that perceives the station as being more powerful than its competition. The same subliminal psychology holds in netcasting too.

Figure 1-1 on page 1-14 shows a 15-minute snapshot of program audio as it emerged from the on-air mixer of a major Los Angeles radio station. Source material included music, speech, and commercials. Notice the large inconsistency in peak and average level between one program source and the next. Figure 1-2 shows the same material after being processed through OPTIMOD-PCn. Notice that program levels are now consistent from source to source

Audio Processing for Netcasts

Professional-grade netcasting requires audio processing similar to FM broadcast, although there are some important differences in the peak limiting because of the different characteristics of the pre-emphasized FM channel and the perceptually coded netcasting channel. In particular, netcasting to mobile devices benefits from audio processing to overcome external noise.

Your listeners deserve to get the best quality and consistency you can provide. Good audio processing is one important thing that separates the amateur from the professional.

Conventional AM, FM, or TV audio processors that employ preemphasis/deemphasis and/or clipping peak limiters do not work well with perceptual audio coders such as Modulation Index’s SteamS MPEG-4 AAC/HE-AACv2 streaming encoder. The preemphasis/deemphasis limiting in these processors unnecessarily limits high frequency headroom. Further, their clipping limiters create high frequency components—distortion—that the perceptual audio coders would otherwise not encode. None of these devices have the full set of audio and control features found in OPTIMOD-PCn.

Peak clipping sounds bad even in uncompressed digital channels because these channels do not rely on preemphasis/deemphasis to reduce audible distortion. Instead of peak clipping, OPTIMOD-PCn uses its advanced MX limiting technology (which includes look-ahead elements) to protect the following channel from peak overload.

About Target Loudness and ITU-R BS.1770

OPTIMOD-PCn uses the contemporary concept of “target loudness” to increase listener satisfaction by minimizing the need for listeners to readjust their volume controls when changing between different broadcast stations or netcast streams. If you specify the target loudness (via the TARGET LOUDNESS control), OPTIMOD-PCn will produce the desired loudness.

The following text explains the concept of "target loudness" and its relationship to the ITU-R BS.1770 loudness measurement algorithm. Instructions for setting target loudness start on page 1-20.

The ITU-R BS.1770 Loudness Measurement Algorithm

In 2006, the ITU-R published Recommendation ITU-R BS.1770: "Algorithms to measure audio programme loudness and true-peak audio level." In 2011, this was updated to BS.1770-3.

> The original BS.1770 loudness meter used a frequency-weighted RMS measurement intended to be integrated over several seconds — perhaps as long as an entire program segment. As such, it is considered a "long-term" loudness measurement.
>
> A major disadvantage of the BS.1770-1 meter was that it weights silence and low-loudness material the same as high loudness material. This will cause the meter to under-read program material (like dialog) having substantial pauses that contain only low-level ambience, because louder program material contributes most to a listener's perception of overall program loudness.
>
> To address this problem, the BS.1770-2 algorithm (and its successors) use gating. There are two steps in the gating process: first, an absolute gate removes silent passages; second, a relative gate weights louder parts of the program more heavily than quieter parts. The gating causes the meter to ignore silence and to integrate only program material whose loudness falls within a floating window extending from the loudest sounds within the specified integration period to sounds that are 10 dB quieter than the loudest sounds.

OPTIMOD-PCn's built-in BS.1770 loudness meter computes gated, integrated loudness over both a 3-second and 10-second rolling window of integration, so all program material from the current time to 3 or 10 seconds before the current time is

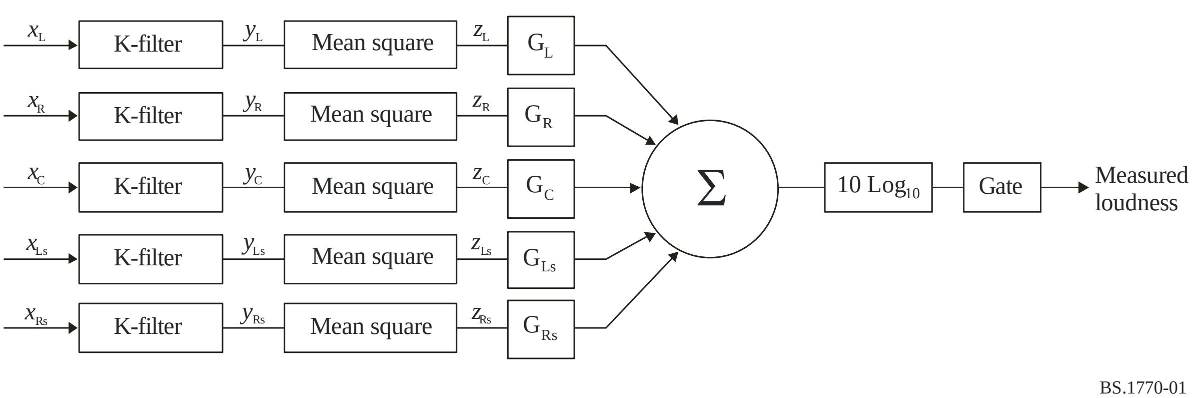

Figure 1-3: BS.1770 Loudness Meter Block Diagram (from ITU-R document)

weighted equally. The 10-second loudness is displayed as a blue bar and the 3-second loudness is displayed as a single white segment.

The meter is scaled so that "0" corresponds to the active TARGET LOUDNESS value. Hence, the loudness level at the consumer's receiver will be correct when OPTIMOD-PCn's processing is adjusted to make the dominant program material indicate "0 LU" on BS.1770 meter and the active Target Loudness value is the same as the receiver's target loudness. (The CBS meter will usually peak close to "0 dB" too; see page 1-26 for a discussion of the CBS meter.) Because loudness perception combines the contributions of all acoustic sources, there is only one Loudness Level meter indication regardless of the number of audio channels

Target Loudness (aka "dialnorm")

The arrival of the ITU-R BS.1770 loudness measurement standard has changed industry practices by encouraging users to specify "target loudness" based upon this standard. Target loudness is also known as "dialnorm," particularly in sound-for-picture applications.

> dialnorm is short for "dialog normalization," and was first specified by Dolby Laboratories (long before the creation of BS.1770) as part of the metadata for their AC3 (later, "Dolby Digital") codec. dialnorm was originally intended to allow motion pictures to be exhibited theatrically with fixed and predictable dialog loudness while allowing other program elements to be louder or quieter than dialog according to the artistic demands of the material. Dolby has since updated their recommended loudness measurement practices to use the BS.1770 standard.

Several national and international standards-setting bodies have incorporated BS.1770 into recommended practices. The advantage of enforcing target loudness is that it causes the loudness of various program elements in a broadcast or netcast to be roughly consistent with each other, so that it is less likely that users will need to adjust their volume controls between program segments. The eventual goal is to achieve consistent loudness across all media, like FM radio and netcasts, supported by a given receiver or player device.

In 2008, Optimods first incorporated the concept of target loudness in the context of processing of sound-for-picture and allowed it to be defeated for audio-only applications. By specifying a global TARGET LOUDNESS value, the user can be sure that any processing preset automatically produces the desired target loudness.

For the sake of consistency and because the world is moving in the direction of specifying target loudness for all deliverables, OPTIMOD-PCn incorporates the concept of target loudness in all audio processing—it's in the processor's "DNA." OPTIMOD-PCn does this in two ways:

- It allows the target loudness to be assigned to each processing preset individually. This allows "radio-style" presets to be designed in the traditional way by balancing loudness against distortion, as loudness is specified by a given preset's local TARGET LOUDNESS value. Most "radio-style" factory presets have a target loudness of −8 LUFS[6] (about 5 dB quieter than the loudest, most distorted "hypercompressed" CDs), although this can easily be adjusted to suit your goals.

- It allows the target loudness to be assigned globally (as part of the system parameters) so that it applies to any preset whose Target Loudness value is set to GLOBAL. When you edit a processing preset so that its Target Loudness value is GLOBAL, this causes the loudness produced by the preset to be the same as all other presets with GLOBAL Target Loudness. All of the 1600's sound-for-picture presets have GLOBAL Target Loudness, and all radio-style presets can be easily edited to have GLOBAL Target Loudness.

The 1600's loudness meters indicate loudness relative to the active target loudness, such that "0" on the meter corresponds to the target loudness.

EBU R128

In August 2010, the EBU published its Loudness Recommendation EBU R128. It specifies how broadcasters and netcasters can measure and normalize audio using Loudness meters instead of Peak Meters (PPMs) or VU meters only, as has been common practice.

EBU R128 is the result of two years of intense work by the audio experts in the EBU PLOUD Group. The new Recommendation is accompanied by a Loudness Metering specification (EBU Tech 3341), a Loudness Range descriptor (EBU Tech 3342), Loudness test material (various different sequences) Production Guidelines (EBU Tech 3343) and Distribution Guidelines (EBU Tech 3344). An EBU Technical Review Article describing the fundamental change in audio in broadcasting is also available: *On the way to Loudness Nirvana*.

EBU R128 recommends normalizing audio at −23 LUFS ±0.5 LU (±1 LU for live programs), measured using the BS.1770-2 (gated) algorithm or higher. The metering approach can be used with virtually all material. To make sure meters from different manufacturers provide the same reading, EBU Tech 3341 specifies the 'EBU Mode', which includes a Momentary (400 ms), Short term (3s) and Integrated (from start to stop) meter.

> In our opinion, −23 LUFS is too low for certain applications. For example, as of 2015 streams normalized to −23 LUFS cannot produce satisfying listening levels on Apple iPhones because the range of the iPhone's volume control is insufficient. We expect that consumer electronics manufacturers will correct this problem in the future, but a decision to set a particular target loudness should also take into account the presence of legacy devices. We believe that a more realistic target loudness for current play-

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[6] The BS.1770 loudness meter indicates in units of LUFS or LkFS (which are identical; the LUFS nomenclature is typically used in Europe and LkFS elsewhere). A change of 1 LUFS is the same as a change of 1 dB. These measurements are absolute with reference to digital full-scale. LU or Lk (without the "FS") refers to loudness relative to the Target Loudness, where the Target Loudness is 0 LU or 0 Lk.

er devices like iPhones is −16 LUFS. This allows very high subjective quality while also allowing the program to be played at a level that satisfies listeners.

ATSC A/85

In 2009, the Advanced Television Systems Committee (ATSC) in the United States released a Recommended Practice: *Techniques for Establishing and Maintaining Audio Loudness for Digital Television* (A/85:2009). This was later updated as A/85:2013. A/85 specifies use of the latest version of the Integrated ITU BS.1770 algorithm for measuring the loudness of DTV broadcasts.

In December 2011, the FCC adopted rules implementing the CALM Act[7], which, by law, forbids commercials from being louder than non-commercial program material. The new FCC rules incorporated ATSC A/85 (and, by implication, the BS.1770 meter) as an objective means of verifying that the rule was being obeyed.

The most important difference between R128 and A/85 is that R128 recommends that the target loudness should be measured across all program material, while A/85 recommends that it be measured on the "anchor element," which is usually dialog. In addition, R128 suggests a target loudness of −23 LUFS, while A/85 suggests a target loudness of −24 LUFS.

The ATSC A/85, ITU-R BS.1770, and EBU R128 documents are available as free downloads and their current versions can easily be located with a search engine.

Setting Loudness

To make automatic loudness control as straightforward and dependable as possible, OPTIMOD-PCn operates somewhat differently from other Optimods. Most important, adjusting the output level control *does not change loudness;* it only sets the amount of headroom between 0 dBFS and the maximum peak output level that the audio processing produces. This allows you to adjust the processing to compensate for downstream overshoots from codecs *without changing loudness*. (Instead, the processing produces more peak limiting.) For example, for the HE-AAC codec it is wise to allow 1.5 dB of peak headroom by setting the output level control to −1.5 dBFS.

- Target Loudness/dialnorm: OPTIMOD-PCn works very easily with Dolby Digital® transmission systems and other systems having a specified target BS.1770 loudness if you do one crucial thing: You must tell the OPTIMOD-PCn what value of Dolby Digital dialnorm metadata (which is the same as BS.1770 target loudness) you are transmitting to your audience, or the target loudness of the receiver if it is not metadata-aware. This will prevent your transmission from being too loud or quiet compared to other correctly set up transmissions.

[7] The CALM Act applies only to U.S. broadcasters. cable providers, and satellite providers.

In a bitstream that includes dialnorm metadata, setting TARGET LOUDNESS to a less negative value automatically turns down the home receiver's volume control, so OPTIMOD-PCn's output level must be turned up by the same amount to maintain a constant loudness at the receiver. Because it is placed before OPTIMOD-PCn's look-ahead limiter, OPTIMOD-PCn's hidden Target Loudness gain control achieves this while allowing OPTIMOD-PCn's look-ahead limiter to prevent digital clipping in the downstream transmission chain regardless of OPTIMOD-PCn's Target Loudness setting. This arrangement allows the user to set the correct loudness at OPTIMOD-PCn's output solely by adjusting OPTIMOD-PCn's active Target Loudness value—it is unnecessary to adjust any other controls within a factory processing preset.

For recommendations regarding player devices that are not Target Loudness-aware, see EBU − TECH 3344: *Practical guidelines for distribution systems in accordance with EBU R128*. The document can be easily found with an Internet search engine.

- Setting Output Loudness: To set the OPTIMOD-PCn's output loudness, adjust its TARGET LOUDNESS value (either within the active processing preset or globally), or adjust the MB LIMITER DRIVE control in the active processing preset.
  - Adjusting TARGET LOUDNESS changes output loudness without changing the indication on OPTIMOD-PCn's loudness meters (which show loudness relative to the active TARGET LOUDNESS value) or the amount of gain reduction in the loudness controller, if turned on. The peak limiter's gain reduction will change. This is the preferred method if the OPTIMOD-PCn's loudness controller is active because it has the smallest effect on the sonic texture of the

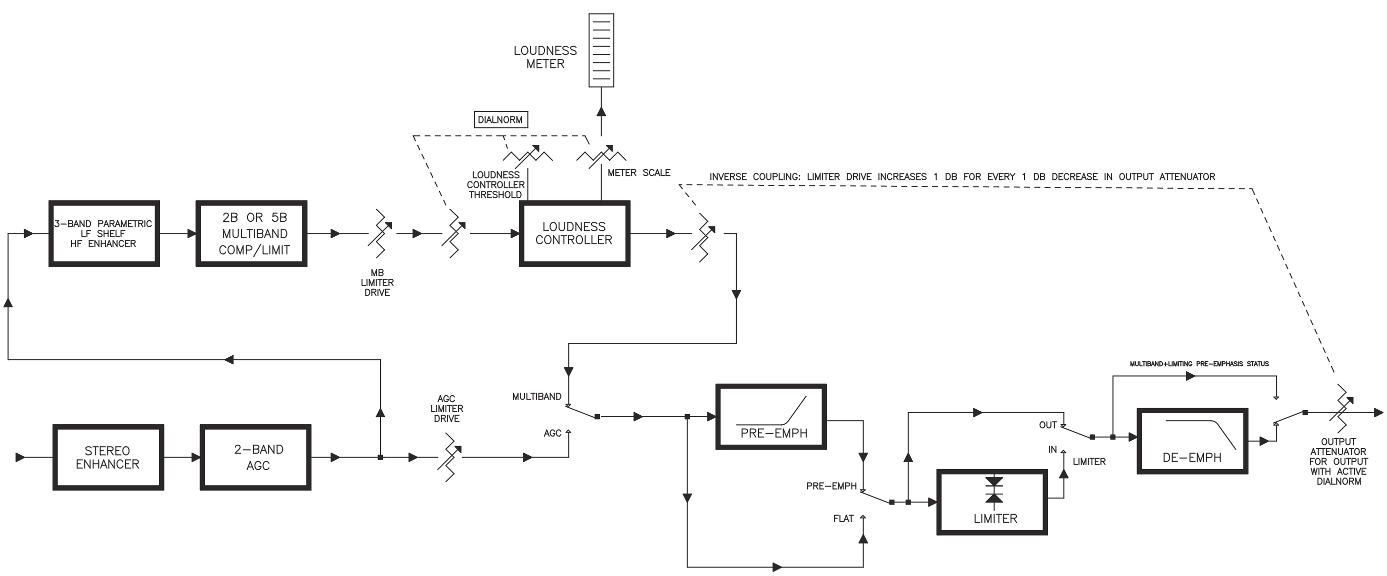

Figure 1-4: Simplified Block Diagram of Target Loudness Control

OPTIMOD-PCn's audio processing.

- Adjusting MB LIMITER DRIVE changes the loudness meters' indications and the amount of gain reduction in the loudness controller and peak limiter. If OPTIMOD-PCn's loudness controllers are turned off (as they are in most radio-style presets), you may have to tweak the MB LIMITER DRIVE to fine-tune the loudness to your program material, such that OPTIMOD-PCn's loudness meters peak around "0."

- Your transmission's loudness will automatically be correct if:
  - The LOUDNESS THRESHOLD and BS.1770 SAFETY LIMITER THRESHOLD controls are set to "0." (If these are OFF, you may have to tweak the MB LIMITER DRIVE control as explained above.)
  - you have adjusted the OPTIMOD-PCn's input reference level (in I/O > INPUT) so that the processing operates with normal amounts of gain reduction (typically 10 dB of AGC gain reduction) and;
  - you have adjusted the OPTIMOD-PCn's TARGET LOUDNESS to match the dialnorm metadata you are send to your audience or the target loudness on player devices that do not accept dialnorm metadata.

The OPTIMOD-PCn's TARGET LOUDNESS value can be set in two places: There is a global setting in the active I/O Setup, which can be overridden by a setting in LESS-MORE tab of the active processing preset. All sound-for-picture factory processing presets are configured to use the global Target Loudness setting specified in the active Setup.

All "radio-style" processing presets have local TARGET LOUDNESS settings, which makes their loudness/distortion/punch tradeoff independent of the setting of global TARGET LOUDNESS. You can make any "radio-style" preset produce the global target loudness by setting the preset's local TARGET LOUDNESS value to GLOBAL and then saving the result as a User Preset. Note that a given preset's LESS-MORE control continues to be available after you have changed the preset's Target Loudness.

To better match other streams in "radio-style" applications, you may wish to set a target loudness that is substantially higher than −23 LUFS. If you see a substantial amount of gain reduction on OPTIMOD-PCn's LIMITER GR meters, we recommend activating the MX limiter (in the DISTORTION tab of the active processing preset). This can substantially improve subjectively quality by reducing or eliminating audible peak-limiting artifacts. However, it will approximately double the CPU usage for that Processor.

If you are using a mono-mode Processor, its BS.1770 and CBS loudness meters will read about 3 dB lower than they do in stereo mode. This is because there is only one audio channel contributing to the loudness measurement. However, the mono output is duplicated on the left and right channels of the output sound device, and if the mono signal is played out through both channels, this will increase the loudness by 3 dB. You must use your best judgment as to how to set up loudness in mono; it depends on listening context.

- CBS Loudness Controller: OPTIMOD-PCn includes third-generation CBS Loudness Controllers™ for DTV applications. These use the Jones and Torick loudness meter as a reference for the loudness controller sidechain. In surround mode, separate loudness controllers are available in the surround and downmix processing chains. (*See Level and Subjective Loudness Metering in OPTIMOD-PCn* on page 1-25 and *Appendix A: Using the ITU BS.1770 and CBS Loudness Meters to Measure Loudness* Controller Performance starting on page 3-93.)
- BS.1770 Safety Limiter: The BS.1770 safety limiter follows the CBS Loudness Controller and will prevent a BS.1770-2 (or higher) loudness meter with a 10-second integration time from indicating higher loudness than the setting of the BS.1770 THRESHOLD control. This limiter is active in all "TV" factory presets. See *BS.1770 Safety Limiter* on page 3-26 for a discussion of why we included this limiter and why we don't think you should use it.
- Customizing a Factory Preset: If you are processing for a distribution channel that has a BS.1770 target loudness and you wish to customize a factory preset while retaining its loudness, you can do this easily. See *Customizing Processing Presets to Achieve Specified Target Loudness* starting on page 2-23 for a more detailed customization procedure for video-oriented presets.

Measuring Studio and Transmission Levels

VU and PPM metering

Studio equipment (like mixers) and transmission equipment (like codecs) typically use different methods of metering to display audio levels. The VU meter is an average-responding meter (measuring the approximate RMS level) with a 300ms rise time and decay time; the VU indication usually under-indicates the true peak level by 8 to 14 dB. The Peak Program Meter (PPM) indicates a level between RMS and the actual peak. The PPM has an attack time of 10ms, slow enough to cause the meter to ignore narrow peaks and under-indicate the true peak level by 5 dB or more. The absolute peak-sensing meter (the type most common in codecs) shows the true peak level. It has an instantaneous attack time, and a release time slow enough to allow the engineer to read the peak level easily. All of OPTIMOD-PCn's level meters are absolute peak sensing.

> Orban offers a free Loudness Meter application for Windows that incorporates a true peak-sensing meter, a VU meter, a PPM, and two types of subjective loudness meters. It can be downloaded from www.orban.com/orban/meter/.

Figure 1-6 shows the relative difference between the absolute peak level and the indications of a VU meter and a PPM for a few seconds of music program.

The studio engineer is primarily concerned with calibrating the equipment to provide the required input level for proper operation of each device, and so that all devices operate with the same input and output levels. This facilitates patching devices in and out without recalibration and ensures that no part of the program chain will clip the audio.

For line-up, the studio engineer uses a calibration tone at a studio standard level, commonly called line-up level, reference level, or operating level. Metering at the studio is by a VU meter or PPM. As discussed above, the VU or PPM indication under-indicates the true peak level. So the studio standardizes on a maximum program indication on the meter that is lower than the clipping level, so those peaks that the meter does not indicate will not be clipped. Line-up level is usually at this same maximum meter indication.

In facilities that use VU meters, this level is usually at 0 VU, which corresponds to the studio standard level, typically +4 dBu. For facilities using +4 dBu standard level, instantaneous peaks can reach +18 dBu or higher (particularly if the operator over-

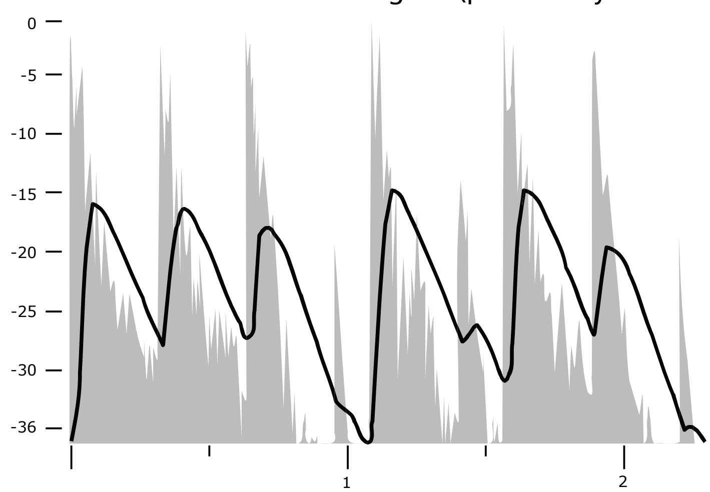

Figure 1-5: VU Meter Indication Compared to Peak Level vs. Time

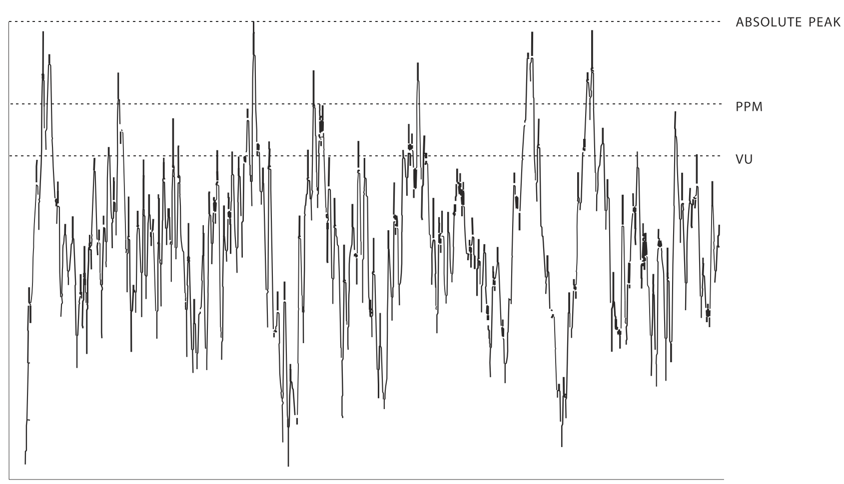

Figure 1-6: Absolute Peak Level, VU and PPM Indications

drives the mixer). OPTIMOD-PCn's analog input clips at an instantaneous peak level of +20 dBu, which provides 16 dB of headroom above a +4 dBu line-up level.

In facilities that use the BBC-standard PPM, maximum program level is usually PPM4 for music and PPM6 for speech. Line-up level is usually PPM4, which corresponds to +4 dBu. Instantaneous peaks will reach +17 dBu or more on voice. In facilities that use PPMs that indicate level directly in dBu, maximum program and line-up level is often +6 dBu. Instantaneous peaks will reach +11 dBu or more.

Figure 1-7 on page 1-25 shows various common meter scales (true peak reading, VU, and four variants of PPM), aligned to show their readings when a −20 dBFS line-up tone is applied to them.

Level and Subjective Loudness Metering in OPTIMOD-PCn

The meters on OPTIMOD-PCn show peak input levels, the peak output modulation, and subjective loudness.

- Input levels are displayed using a VU-type scale (0 to −40dB), but the metering indicates *absolute instantaneous peak* (much faster than a standard PPM or VU meter). The maximum digital word at the input corresponds to the 0 dB point on the 1600's input meter.
- The output meter indicates the values of the digital samples at OPTIMOD-PCn's digital outputs, taking into account the setting of OPTIMOD-PCn's output level control.

Note that this meter does not indicate the reconstructed peak level (aka "0 dBFS+") per BS.1770. The "Reconstructed Peak" meter in Orban's free loudness meter application (www.orban.com/meter) can be used for this purpose.

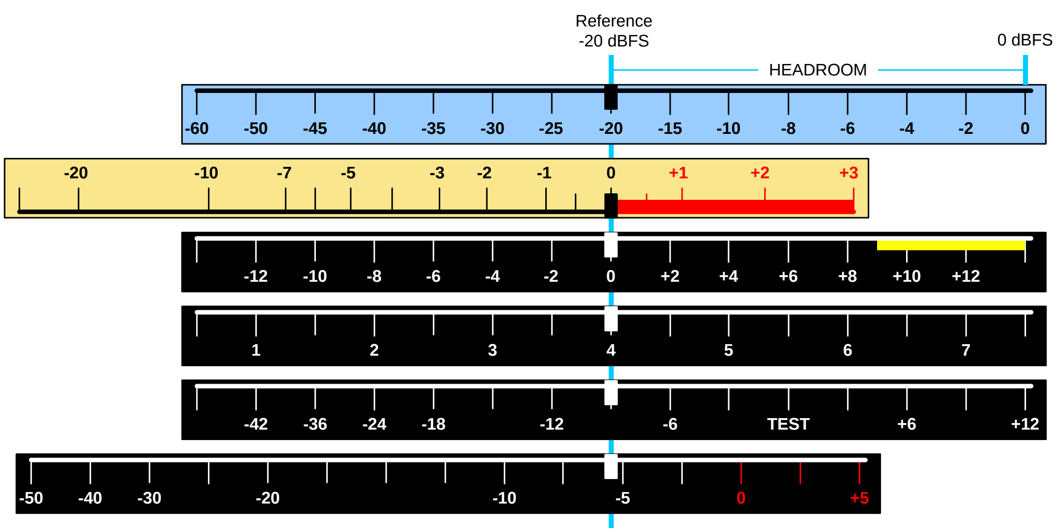

Figure 1-7: Common Audio Meter Scales, Aligned to the Same Reference Level

Your Optimod's base sample rate is 48 kHz; its peak limiters are oversampled at 192 kHz or higher. If the output sample rate is converted to a rate other than 48 kHz and/or passed through a D/A converter, the peak level of the output may increase up to 0.2 dB because 192 kHz is not quite an "infinite" sample frequency, which is what would be required to perfectly predict and control the peak level in these cases.

- The subjective loudness meters, labeled BS1770 and CBS, display loudness measured by the ITU-R BS.1770-3 standard (see *The ITU-R BS.1770 Loudness Measurement Algorithm* on page 1-17) and loudness measured using the CBS Technology Center algorithm developed by Jones and Torick[8], which is discussed immediately below. The loudness meters indicate loudness relative to the active TARGET LOUDNESS value, as set either globally or in the active processing preset (see *Target Loudness* on page 1-18).

The CBS meter is a "short-term" loudness level meter that displays the details of moment-to-moment loudness with dynamics slightly faster than a VU meter. It can indicate the loudness of short-term sounds (like pistol shots) that may be annoying to TV viewers but that the BS.1770 meter, because of its longer integration time, may not take fully into account. Created using proprietary modeling software, the DSP implementation typically matches the original CBS analog meter within 0.5 dB on sinewaves, tone bursts and noise.

The Jones & Torick algorithm improves upon the original loudness measurement algorithm developed by CBS researchers in 1967. Its foundation is psychoacoustic studies done at CBS Laboratories over a two year period by Torick and the late Benjamin Bauer, who built on S. S. Stevens' '50s-era work at Harvard University.

After surveying existing equal-loudness contour curves (like the famous Fletcher-Munson set) and finding them inapplicable to measuring the loudness of broadcasts, Bauer and Torick organized listening tests that resulted in a new set of equal-loudness curves based on octave-wide noise reproduced by calibrated loudspeakers in a semireverberant 16 x 14 x 8 room, which is representative of a room in which broadcasts are normally heard. They published this work[9] along with results from other tests whose goal was to model the loudness integration time constants of human hearing. These studies concentrated on the moderate sound levels typically preferred by people listening to broadcasts (60 to 80 phons[10]) and did not attempt to characterize loudness perception at very low and high levels.

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[8] Jones, Bronwyn L.; Torick, Emil L., "A New Loudness Indicator for Use in Broadcasting," J. SMPTE September 1981, pp. 772-777.

[9] Benjamin B. Bauer and Emil L. Torick, "Researches in Loudness Measurement," IEEE Transactions on Audio and Electroacoustics, Volume AU-14, Number 3, September 1966, pp. 141-151

[10] The phon is a unit of perceived loudness, equal in number to the intensity in decibels of a 1 kHz tone judged to be as loud as the sound being measured.

According to this research and its predecessors, the four most important factors that correlate to the subjective loudness of broadcasts are these:

1. The power of the sound.

2. The spectral distribution of the power. The ear's sensitivity depends strongly on frequency. It is most sensitive to frequencies between 2 and 8 kHz. Sensitivity falls off fastest below 200 Hz.

3. Whether the power is concentrated in a wide or narrow bandwidth. For a given total sound power, the sound becomes louder as the power is spread over a larger number of *critical bands* (about 1/3 octave). This is called *loudness summation*.

4. Temporal integration: As its duration increases, a sound at a given level appears progressively louder until its duration exceeds about 200 milliseconds, at which point no further loudness increase occurs.

Bauer and Torick used the results of this research to create a loudness level meter with eight octave-wide filters, each of which covers three critical bands. (B & T did not use one filter per critical band because this would have made the meter, which was realized using analog circuitry, prohibitively expensive.) Each filter feeds a full-wave rectifier and each rectifier feeds a nonlinear lowpass filter that has a 10 ms attack time and a 200 ms release time, somewhat like the sidechain filter in an AGC. This models the "instantaneous loudness" perception mechanism in the ear. Instantaneous loudness is not perceived directly but is an essential part of the total loudness model.

To map the instantaneous loudness to perceived short-term loudness, the outputs of each of the nonlinear lowpass filters are arithmetically summed with gains chosen to follow the 70 phon equal-loudness curves of the ear as determined by Bauer and Torick's research. The sum is applied to a second, slower nonlinear lowpass filter. This has an attack time of 120 ms and a release time of 730 ms. Along with the eight nonlinear lowpass filters following the individual filters, this filter models temporal integration and maps it to the visual display. Meanwhile, the arithmetic addition models loudness summation.

The internationally accepted unit of subjective loudness is the *sone.* With a sinewave, 40 phons = 1 sone. A doubling of sones corresponds to a doubling of loudness. However, because broadcasters were accustomed to working in decibel units, Jones and Torick chose to map loudness on a LED ladder display encompassing −20 to +5 dB in 0.5 dB increments, with the understanding that the perceived loudness doubles every 10 dB at loudness levels typically heard by broadcast audiences.

The J & T meter is monophonic. Psychoacoustic studies indicate that when multiple acoustic sources are present in a room, loudness is most accurately expressed by summing the power in the sources. For example, driving two loudspeakers with identical program produces 3 dB higher loudness than a single speaker produces. Therefore, to extend the J & T algorithm to multichannel reproduction, we implement one eight-filter filterbank for each channel and compute RMS sums of the outputs of corresponding filters in each channel before these sums are applied to the eight nonlinear lowpass filters. As in the monophonic J & T algorithm, the sum of these lowpass filters drives a second nonlinear filter, which drives the display.

Presets in OPTIMOD-PCn

There are two kinds of presets in OPTIMOD-PCn: Processing Presets and Setup Presets. Both can be customized and saved under different names.

> Each instance of the processing is identified by a Processor number or user-generated alias and each Processor "owns" its individual customized presets.

Processing Presets

There are over 50 Factory Processing Presets, written by Greg Ogonowski and Bob Orban. They are designed to be compatible with almost any program format. See *Table 3-1: Protection, AGC and Mastering* Presets on page 3-34, Table 3-2 on page 3-38, and *Table 3-3: Sound-For-Picture Presets* on page 3-42. The description of the presets associates with these tables indicates the style of processing.

Each Factory Processing Preset on the Open Preset list is really a library of 19 separate presets, selected by using the LESS-MORE control to adjust OPTIMOD-PCn for less or more processing.

> Factory Processing Presets are stored as text files on the hard drive of the same computer that runs the OPTIMOD-PCn Service. During installation of the OPTIMOD-PCn Service, a Presets folder containing all the factory Preset files is created in the `C:\Program Files (x86)\Orban\Optimod-PCn 1600\presets\` folder.
>
> Each set of Factory Preset files consists on one "master" file and several "less/more" files. Master files contain the preset data that is first loaded when you activate a factory preset. Less/more files contain the preset data that is called up when you edit a factory preset via the Control application's one-knob "less/more" editing procedure. If there is no less/more file for the specific less-more setting you choose, OPTIMOD-PCn will automatically generate the data by interpolating between the contents of the two nearest less/more files.
>
> The suffix of the master Factory preset files is `orb1600f`. Within the preset folder on your hard drive, there is a corresponding less/more folder named after the master factory preset file. The Less/More files are located in these folders. The file names of the less/more files are [preset name] `LMxxx.orb1600f`, where "xxx" is three numbers, like "080" (which corresponds to a LESS/MORE value of 8.0).
>
> All of these files have the "read-only" attribute to make them inconvenient to erase, even at the operating system level. You cannot erase or overwrite them from the OPTIMOD-PCn Control application. If you erase or modify a factory or less/more file from an external file manager like Windows Explorer (a very unwise thing to do), you will have to reinstall the OPTIMOD-PCn software to regenerate the file unless you have a backup copy of the file elsewhere.
>
> Each factory preset has an associated folder containing all of the less/more files for that preset. The less/more folders are located immedi-

ately below the `presets` folder; each less/more folder bears the name of its associated preset.

You can create custom "factory" presets that have full LESS-MORE functionality. See *Creating Custom "Factory" Presets* on page 3-90.

Customizing Processing Presets

You can change the settings of a Factory Processing Preset, but if you want to preserve your changes, you must then store those settings as a User Preset, which you are free to name as you wish. You can also create User presets by editing existing user presets and saving the results under a new name.

The suffix of User Presets is `ORB1600USER`. The Factory preset remains unchanged.

You cannot create User Presets from scratch. Start by recalling a Factory preset. You can then immediately store this in a new User Preset (with "Save As" from the FILE menu), give it whatever name you wish, make changes to the settings as desired, and then save it again. Alternatively, you can recall a Factory preset, make the changes first, and then store this as a User Preset. Either way, the Factory preset remains for you to return to if you wish.

You can also modify an existing User Preset.

When you modify an existing preset, whether Factory or User, the OPTIMOD-PCn server software will automatically generate a temporary User Preset whose name consists of "Modified" appended to the front of the existing preset name. If you do not save your modifications, this temporary preset will remain on the server's hard drive until you further modify any preset. Then the temporary preset will be overwritten.

You can store as many User Presets as the 1600PC application's host computer hard drive and operating system can accommodate. User Presets are shown on the "Open Preset" list by the name that you gave them when you saved them.

You can name them as you wish, limited only by the file naming limits in your operating system.

Do not use a suffix; .`orb1600user` will be added automatically.

User Presets are not stored the `C:\Program Files (x86)\Orban\Optimod-PCn 1600\presets\` folder. By default, they are instead stored in on the computer running the OPTIMOD-PCn Service in

`C:\Users\Public\Documents\Orban\Optimod-PCn 1600\Processor[x]\presets`,

where "Processor[x]" is the name of the audio Processor (for example, "Processor1"). Additionally, when it connects to the OPTIMOD-PCn Service, 1600PC makes copies of the Factory Presets and store them here too (although without their associated LESS-MORE folders).

If you are running 1600PC on a different computer from that running the OPTIMOD-PCn Service, 1600PC automatically copies to the remote computer the User Presets associated with the audio Processor to which you are connected. It also copies the Factory Presets. Again, the default folder is

`C:\Users\Public\Documents\Orban\Optimod-PCn 1600\Processor[x]\presets`.

A User Preset that you create while working with a given Processor is available only to that Processor. If you wish to copy User Presets from Processor x to Processor y, you must use Windows to copy the corresponding *.ORB1600USER file from

`C:\Users\Public\Documents\Orban\Optimod-PCn 1600\Processor[x]\presets` to

`C:\Users\Public\Documents\Orban\Optimod-PCn 1600\Processor[y]\presets`.

Do this on the computer running the OPTIMOD-PCn Service.

If you want to back up a user preset, use the standard Windows file copy mechanism to copy it from any folder where it is currently located into a backup folder you have made. It is wise to back up User Presets regularly.

If you want to delete a user preset, use Windows to delete the preset's associated `*.ORB1600USER` file.

Setup presets

Setup presets contain setup information, such as input levels, output levels, global TARGET LOUDNESS, and operate/defeat switches for various signal-processing blocks. Like Processing presets, you may customize Setup presets and save them from FILE > SAVE SETTINGS AS in the control application.

Overview of an OPTIMOD-PCn Installation

OPTIMOD-PCn 1600 is a Windows Service running on Microsoft Windows 7 or higher and an Intel x86 CPU. All encoding and audio processing is performed natively on the x86 CPU.

The 1600's OPTIMOD-PCn audio processing is controlled via the 1600PC control application, which can run locally or remotely.

The 1600PC control software is supported only on Windows 7 and higher computers.

The OPTIMOD-PCn 1600 Service can spawn up to 16 independent audio processors, depending on the number of Processors you have purchased. Each Processor can be controlled independently by the 1600PC control application.

> The number of Processors you have purchased determines the maximum number of instances permitted on your system. However, turning on CPU-intensive parts of the OPTIMOD-PCn 1600 processing, like the MX limiter or Optimix upmixer, may limit the number of available instances to fewer than the maximum because all of the CPU power has been used up. If the CPU is overloaded, it will cause audio glitches and stuttering. See Table 1-1 on page 1-5 and associated text.

Operationally, Processors are almost independent of each other. 1600PC connects to only one Processor at a time, allowing you to recall, edit and save presets, set input/output mix levels, and do other housekeeping tasks for that Processor. Multiple instances of 1600PC can be connected simultaneously to different Processors, regardless of whether they were spawned by a single OPTIMOD-PCn Service or by multiple Services on multiple computers.

An important exception to this principle of independence is that changing the setup parameters of a given Processor, like its input and output sound devices, requires the Service to be stopped and restarted. *This will briefly interrupt audio for all Processors* and will happen automatically from the TOOLS > SERVICE SETTINGS menu if you open it and then click OK. *If you accidentally open the Service Settings menu and do not want to interrupt audio, click CANCEL.*

Processors are named "Processor1," Processor 2," etc., and are identified as such in the Service Settings window in 1600PC. For convenience, you may assign each one a name ("alias"). Instructions for doing so are provided in Section 2 of this manual. This allows the 1600PC control application to be connected to the desired Processor and allows you to specify the audio input and output routing of each instance.

Conceptually, there are two options for operating OPTIMOD-PCn: *local* and *network.* Regardless of whether operation is "local" or "network," a server that accepts TCP/IP communications runs in the background on the 1600.

- *Network* operation allows you to control one or more OPTIMOD-PCn Services over a network, regardless of whether the Service is located in the controlling computer. This is the recommended configuration. It allows the computer running the Service to provide a maximum number of audio Processors and facilitates locating this computer in a machine room without being connected to a keyboard, mouse, or monitor.
- *Local* operation runs the 1600PC control application on the same computer as the Service. Local operation is not preferred because it dedicates some of the 1600's CPU power to running the remote software instead of audio processing or encoding, and because the computer running the Service may not be equipped with high-resolution graphics.

The 1600PC control application always uses TCP/IP to communicate through the 1600's server software to instances of the audio processing. All Processors running in one 1600 Service are controlled via the same IP address and port numbers.

Start up the 1600PC control application as you would any Windows application. As of this writing, it is necessary to run 1600PC with Administrator privileges so that it can address the parts of the Registry that contain the Service parameters.

The Control application will initialize without connecting to an instance, so you must connect the application to the instance you wish to control.

Networking the 1600: Overview

The 1600PC control application can control one or more 1600s installed anywhere on a TCP/IP network, including the Internet. A complex system (such as one that a major broadcast group, or large server farm/ISP might operate) could have dozens of clients and servers networked together. Each host computer has a TCP/IP address and port assigned to it and runs one instance of the OPTIMOD-PCn Service. We will call such a host computer a "server." The Service is the "switchboard" that allows the 1600PC to control multiple Processors running on one computer.

The control application can automatically find all Processors within a given host computer anywhere on the network if the control application knows the IP address of the host computer and the Port number assigned to the Service in TOOLS > SERVICE SETTINGS.

To connect to a Processor, you must first add the host computer/server to your control application via CONNECT > ADD PROFILE (step 15 on page 2-12). Here, you provide the host computer's IP address and port number. Once a profile is added, you can connect to it by choosing the desired Processor in the CONNECT drop-down list.

Each local copy of the Optimod-PCn Control application allows you to name a 1600 server (e.g., "Server 1") so you do not have to remember its IP address each time you connect the Control application to it.

> The server names are stored locally on each computer; each local copy of the Optimod-PCn Control application can create a different profile name for a given server. A server's real "name" is always its IP address.

You can see a drop-down list of all instances that have been added previously, regardless of whether you are logged onto the host computer housing those instances. Once you have logged onto the host computer, you can connect to any Processor on this list if it is "available" (that is, if it running in the host). If you have not previously connected to a given instance and specified "remember password," you must supply your User Password before you can connect.

> Once you have specified "Remember Password," the password dialog box will not appear again and the "Remember Password" status can only be canceled by an administrator. "Remember Password" can compromise security, so you should use this feature with discretion.

Only one client can be logged into a given Processor at one time; the instance will return “in use” if another client attempts to log into the same instance. However, more than one client can be logged onto a given host computer at once if each client is logged onto a different Processor within that host.

Section 2 of this manual provides detailed, systematic instructions for setting up a network.

Security

OPTIMOD-PCn is designed for networking. Because most PCs are now networked, the instances must be protected from unauthorized access when networking is activated. Two levels of security achieve this:

- Each Processor has a User password that allows an authorized user to “connect” to the instance via the OPTIMOD-PCn Control Application. When this occurs, the user can work with the audio processing functions in the instance, can change and edit presets, and can do other tasks similar to those that one would do on a stand-alone audio processor. Additionally, each Processor has a Terminal Password (operating at the User security level) that allows you to connect to an instance via the OPTIMOD-PCn TCP/IP terminal server to allow external control of the Instances from either a Telnet/SSH client or a custom third party application. See OPTIMOD 1600PCn Control API on page 5-1.
- Each OPTIMOD-PCn Service has an Administrator password that allows an administrator to set up the Service and the various Processors.

    Only an administrator can assign and change user passwords. If the user has checked “remember password” on the dialog box requesting password entry, only the administrator can “uncheck” the “remember password” function for that instance.

    Unless networking is activated, Processors located in a given machine are visible only to the Optimod-PCn Control application running locally on that machine. They are invisible to other machines on the network. Not running in “network” mode therefore provides considerable security.

- By creating a “Network Accept List,” you can specify which computers are permitted to connect to a given OPTIMOD-PCn instance. This provides a layer of security that complements password protection. See step (7.C) on page 2-7.

Each audio Processor is enumerated as “Processor[x],” where x is a digit, with the first instance starting at 1. *The number is the fundamental means by which the Optimod-PCn software identifies the instance*.

The Optimod-PCn software allows you to give a Processor an easily remembered name (e.g., “KORB Streaming”) so the software does not have to address the Processor by its generic name. *This name is an alias that is local to the computer from which you named the Processor; the same Processor could be known by a different name on each computer in the network.*

If you give your Processor an alias name, this name will appear (instead of PROCESSOR 1, etc.) next to the Service name in the Connect drop-down box and at the top left hand corner in the 1600PC application when you are connected to that instance.

> Only the generic name ("Processor[x]") is unique. The name is always available on "Service Settings" screen, which is accessible from the Tools drop-down menu box. The 1600PC Remote application uses the generic name to identify the folder on the host computer's hard drive that contains the setup and user presets (if any) for that instance.
>
> Only an administrator can rename an instance (profile), and that name will remain only on the machine from which it was assigned that name.

For 1600PC to access Processors on other machines, 1600PC must always know these remote Processors' passwords to access them. The administrator initially assigns these passwords at the machine in which the remote Processors are installed. If you plan network operation, it is very important to assign both a User and an Administrator password to each Processor in that machine to prevent the Processors from being hacked.

> The dual password system is useful in protecting a networked installation from being damaged by disgruntled employees or hackers who might get access to a User password and server IP address. A malicious user might set incorrect presets and audio levels, activate test tones, mute the audio, or delete user presets. (This is another reason, other than potential hard drive failure, why it is wise to back up user presets.)
>
> However, a malicious user cannot take exclusive control of a Processor by changing its name or password. Only an administrator can do that. Meanwhile, the administrator can change User passwords and rename Processors from a central location. It is wise to do this each time a person with a User password leaves the employ of the entity doing the streaming or broadcasting—it's like "changing the locks."

No networked Processor is viewable inside 1600PC Remote software until you "add" its Profile (which encompasses all audio Processors spawned from a given computer's Service) to the list of viewable instances by going into Connect>Add and then supplying the IP address and the instance's User password (if a password was assigned to that instance). This allows one host computer to accommodate multiple Processors, each processing competitor's streams, while allowing a given competitor to view only its own Processor(s) from the Control application.

Input/Output Configuration

To move audio into and out of OPTIMOD-PCn, use normal sound input/output hardware having Windows drivers. The drivers must support WASAPI at the output side. OPTIMOD-PCn uses Windows MME on the input side so that Windows can perform sample rate conversion if necessary.

The input/output sample rate of OPTIMOD-PCn is 44.1 kHz, 48 kHz, 96 kHz, and 192 kHz, 16-bit or 24-bit. Using Windows Control Panel > Sound, you must configure the sound devices you are using for the sample rate and bit depth of your choice.

> We recommend 48 kHz, 24-bit. The processing always runs internally at 48 kHz. 44.1 kHz, 96 kHz and 192 kHz support were added for the convenience of users who are embedding the Optimod into facilities built around one of these sample rates. Because the high-quality synchronous sample rate converters in your Optimod put additional load on your CPU, do not use 44.1 kHz, 96 kHz or 192 kHz unless you have a good reason to do so.

Locking Clocks: If the input and output sound devices are not on the same hardware (for example, if they are on two separate soundcards), to achieve long-term stability *the sample rates of the input device and output device must be proactively locked together. Simply setting the sample rates of the input and output device to the same numerical value is insufficient.* Failure to proactively lock sample rates will cause the musical pitch of the output to be slightly different from the input and can cause glitching and/or a delay build-up between input and output because samples are not being read out from the output at the same rate that they are being applied. Synchronization is customarily achieved with a wordclock connection to both devices. If you are using an audio-over-IP connection, be sure that the wordclock generator is locked to the AoIP network sample rate, which is customarily determined by "Grandmaster" clock hardware on the network.

See the diagram on page 1-41 for typical correct (labeled ☑) and incorrect (labeled 🛇) setups with an internal audio source. See page 1-43 for correct and incorrect setups with an external audio source. These diagrams can be magnified as needed.

Subject to the limitations below, you may use Virtual Audio Cable to move audio from one application to another (for example, from a playout system to the input of OPTIMOD-PCn), VAC is available for purchase at a reasonable price[11]. VAC includes a driver that allows you to create multiple "virtual audio cables" in software, so you can interconnect audio applications without looping them through a hardware sound card. Each cable you create appears in Windows' list of available sound I/O devices.

To avoid glitches or delay build-up, the input and output sample rate of a given application must be precisely locked together, but the VAC clock frequency cannot be locked to a hardware sound device. Moreover, older versions of VAC did not support locking multiple VAC "cables" to the same (internal) clock. *Hence VAC can only be used safely with a given application if that application uses VAC for both input and output and you are using the current version of VAC*.

An example of a safe chain within a given computer is:
Playout System > VAC > OPTIMOD-PCn > VAC > Streaming Encoder > Network.

---

[11] http://software.muzychenko.net/eng/vac.htm

You must carefully configure VAC to match how you have set up the input and output sample rate and bit depth (step 10 on page 2-8). The input of a stereo or mono instance of OPTIMOD-PCn is always two-channel. If you have purchased the upmixing version of a stereo Processor, then that Processor can be configured for stereo or 5.1 surround output; 5.1 is needed if you wish to use the Optimix upmixer. If a given Processor is configured for stereo output, the Optimix controls will not appear in 1600PC.

Using OPTIMOD-PCn for Netcasting

It is usually best to run the OPTIMOD-PCn audio processor on the same computer that runs your streaming encoder. This simplifies interconnection between the audio processor(s) and encoder(s). In Windows installations, we recommend using Virtual Audio Cable to pass the processed output of OPTIMOD-PCn to the streaming encoder.

OPTIMOD-PCn's inputs accept uncompressed PCM-format digital audio from a Windows Sound Recording Device, as shown in the Windows Sound Control Panel. For surround processing, the device must be capable of multichannel operation.

The inputs will not accept "bitstream" inputs encoded with formats like Dolby Digital® or DTS®. Inputs must be "PCM" (Pulse-Code Modulation) format with sample rates of 44.1 kHz, 48 kHz, 96 kHz, or 192 kHz and 16 or 24 bit word length.

Using OPTIMOD-PCn in Digital Radio Service

The best location for OPTIMOD-PCn is as close as possible to the transmitter so that OPTIMOD-PCn's digital output can be connected to the transmitter through a signal path that introduces no change in OPTIMOD-PCn's output PCM bitstream.

Sometimes it is impossible to locate OPTIMOD-PCn at the transmitter. Instead, it must be located on the studio side of the link connecting the audio facility to the transmitter. If the transmitter is not accessible, all audio processing must be done at the studio, and you must tolerate any damage that occurs later.

If an uncompressed digital link is available, this is an ideal situation because such a link will pass OPTIMOD-PCn's output with little or no degradation. However, such a link is not always available.

> If only a 32 kHz sample rate link is available, the sample rate conversion necessary to downsample the audio will cause overshoots when OPTIMOD-PCn is operated at 20 kHz bandwidth because the sample rate converter removes spectral energy. In this case, you can minimize overshoot by operating OPTIMOD-PCn at 15 kHz bandwidth. (Set it from the GLOBAL tab in the I/O window.)

Unless the path is a digital path using no lossy compression, this situation will yield lower performance than if OPTIMOD-PCn is connected directly to the transmitter because artifacts that cannot be controlled by OPTIMOD-PCn will be introduced by the link to the transmitter. These artifacts can decrease average modulation by 2-4dB and can also add noise and audible nonlinear distortion. In the case of lossy digital compression, this deterioration will be directly related to the bitrate. For an analog path, the deterioration will depend on the amount of linear and nonlinear distortion in the path.

One strategy is to apply to OPTIMOD-PCn's output signal the same lossy compression that the digital radio transmitter would apply. If a digital link is available with sufficient bitrate to pass this compressed signal, it can then be passed directly to the digital radio transmitter without further processing if synchronization issues can be resolved. Consult with the manufacturer of your digital radio transmitter to see if this can be done.

Where only an analog or lossy digital link is available, feed the audio output of OPTIMOD-PCn directly into the link. If available, the transmitter's protection limiter should be adjusted so that audio is normally just below the threshold of limiting: The transmitter protection limiter should respond only to signals caused by faults or by spurious peaks introduced by imperfections in the link.

Where maximum quality is desired, it is wise to request that all equipment in the signal path after the studio be carefully measured, aligned, and qualified to meet the appropriate standards for bandwidth, distortion group delay and gain stability. Such equipment should be measured at reasonable intervals.

It is important to understand that the codecs used in digital radio service introduce overshoot as a side-effect of bitrate reduction. It is wise to evaluate the codec in use for this issue. In general, allowing 2 dB of headroom (by setting OPTIMOD-PCn output level control at −2 dB) will suffice even for HE-AAC because overshoots above 2 dB have low energy and are unlikely to be audible.

OPTIMOD-PCn at the Transmitter: Gain Control before the STL

The audio received at OPTIMOD-PCn's input should have the highest possible quality. To achieve the full audible benefit of OPTIMOD-PCn processing, use a studio-transmitter link (STL) that is as flat as the bandwidth of OPTIMOD-PCn as used in your facility (usually 20 kHz). Ideally, you should use a 20-bit (or better) uncompressed digital link with at least 44.1 kHz sample frequency.

Because the audio processor controls peaks, it is not important that the audio link feeding OPTIMOD-PCn's input terminals be phase-linear. However, the link should have low noise, the flattest possible frequency response from 20-20,000Hz, and low nonlinear distortion.

If the audio link between the studio and the transmitter is noisy (or, if digital, is limited to 16 bits or less), performing the AGC function at the studio site can minimize the audibility of this noise. AGC applied before the audio link improves the signal-to-noise ratio because the average level on the link will be greater. Further, many STLs require level control to prevent the STL from being overloaded. You can use another 1600 for this purpose.

Setting Output/Modulation Levels

In a perfect world, one could set the peak level at OPTIMOD-PCn's output to 0 dBFS. However, there are at several potential problems that may make it desirable to set the modulation level slightly lower.

IMPORTANT: In OPTIMOD-PCn, adjusting the output level control *does not change loudness;* it only sets the amount of headroom between 0 dBFS and the maximum peak output level produced by the audio processing. This allows you to easily adjust the output level control to compensate for downstream overshoots. See *Setting Loudness* on page 1-20 and step (4.C) on page 2-18.

- First is the fact that the peak limiter operates at a finite sample rate: 192 kHz (non-MX) or 256 kHz (MX). This ensures that overshoot after phase-linear sample rate conversion or D/A conversion will not exceed 0.5 dB and will typically be less than 0.1 dB.
- Second is additional processing, such as equalization. Equalization that applies boosts at certain frequencies is very likely to add peak level and thus cause clipping. However, equalization that attenuates certain frequencies can also cause overshoots because of added phase shifts. So be wary of any equalization and allow headroom to accommodate it.
- Third is headroom in lossy data compression systems. A well-designed perceptual encoder will accept samples up to 0 dBFS and will have internal headroom sufficient to avoid clipping. However, there is no guarantee that *receiver* manufacturers or *decoder* providers will implement perceptual decoders with sufficient headroom to avoid clipping overshoots, and in fact, some very popular player devices and software suffer from clipping in the analog domain due to overshoot. Such overshoots are the inevitable side effect of increasing the quantization noise in the channel, and can be as large as 3-4dB. Most perceptual encoder algorithms are designed to have unity gain from input to output. So if peak levels at the input frequently come up to 0dBFS, peak levels at the output will frequently exceed 0dBFS (and will be clipped) unless the decoder algorithm is adjusted to be less than unity gain.

  Canny engineers will therefore familiarize themselves with the performance of real-world receivers and player software and will reduce the peak modulation of the transmissions if it turns out that most receivers are clipping due to perceptual encoding overshoots. Our experience to date indicates that allowing 3dB headroom should prevent audible overshoot-induced clipping in low bite-rate systems (e.g., 32 kbps streams), while 2dB is adequate for 128kbps and above. While some clipping may still occur, it will have a very low duty cycle and will almost certainly be inaudible.

In sound-for-picture applications, the goal is often to have the loudness of your broadcast or stream be the same as the loudness of other broadcasts or streams so that consumers do not have to readjust their volume controls when changing from one program stream to another. EBU R128 extends this concept to other media and transmission types with the goal of delivering consistent loudness to consumers regardless of the program type or source.

Dolby digital broadcasts and streams contain "dialnorm" metadata to help achieve a specified target loudness. OPTIMOD-PCn 1600 has been designed to always support dialnorm (called "target loudness" in OPTIMOD-PCn), making it easy to ensure correct loudness. While some of the concepts may seem unfamiliar at first, learning about them can pay off by increasing audience comfort and satisfaction. See *About Target Loudness and ITU-R BS.1770* starting on page 1-16.

Managing Delay from Input to Output

To optimize audio quality and to allow adding features like automatic phase skew correction and surround upmixing, we chose to design the 1600 audio processing software without regard to input/output delay. Accordingly, the delay can be as much as 1 second. Because of limitations of the Windows audio system, this delay is not 100% repeatable; it is not possible to achieve sample-accurate delay matching of the 1600's audio processor to an external delay.

However, the differential delay between the surround and downmix outputs of a surround instance is fixed to sample accuracy. Moreover, the delay of a given instance in "pass-through" mode is always matched to its delay in "operate" mode, so it is possible to crossfade between the modes smoothly.

OPTIMOD-PCn allows the functions listed below to be smoothly activated or defeated during audio programming, which is done via controls in the processing presets. Because these functions each add significant input/output delay and CPU usage, Setup presets include switches that can defeat these functions entirely and remove their delay from the audio path.

- Phase Skew Correction
- Stereo Synthesis
- Optimix upmixing

Conversely, the MX limiter functions (MX LIMITER IN/OUT and MX OVERSHOOT SOFT/HARD) change the I/O delay when they are turned on and off. This is because the MX limiter is very CPU-intensive, so it is undesirable to run the MX processing when it is unneeded just to permit smooth switching between MX and non-MX limiting.

Because the I/O delay changes, it is impossible to switch between presets having different settings of the MX LIMITER IN/OUT and/or MX OVERSHOOT SOFT/HARD controls without introducing audio glitches. This is usually not a problem because MX operation always equals or outperforms non-MX operation. (It will outperform whenever a significant amount of peak limiter gain reduction occurs, as indicated on the

LIMITER G/R meters.) The only reasons to defeat MX operation are to decrease CPU loading and/or delay.

Monitoring on Loudspeakers and Headphones

In live operations, highly processed audio often causes a problem with **the DJ or presenter's headphones.** Thanks to no-compromise audio processing, the delay through OPTIMOD-PCn can be as much as 1 second, so its output cannot be used for monitoring in live broadcasting, as it will affect audio cueing and DJ mixing in addition to being unacceptably distracting through headphones.

Such problems can be avoided if the DJ/presenter's headphones are driven directly from the program line or, better, by an inexpensive analog compressor connected to the program line. If the DJ/presenter relies principally on headphones to determine whether a digital radio station is on the air, simple loss-of-data and loss-of-audio alarms should be added to the system. Such alarms could be configured to cut off audio to the DJ/presenter's phones when an audio or carrier failure occurs.

If monitoring a live broadcast though audio processing is essential, one can instead use Orban's OPTIMOD-PC 1101 card, which uses onboard DSP to achieve low delay. However, this card is cannot provide audio quality as high as that delivered by the 1600.

Controlling Levels in a Playout System Feeding OPTIMOD-PCn

To optimize the consistency of a broadcast or netcast, preprocessing each program element via OPTIMOD-PCn before it is stored on a playout system is not as effective as processing the playout system's output in real time using OPTIMOD-PCn. The latter technique maximizes the smoothness of transition between program elements and makes voice from announcers or presenters merge smoothly into the program flow, even if the announcer is talking over music.

You can help OPTIMOD-PCn operate best by setting the level of each program element when you load it into the playout system. Many audio editing programs permit a sound file to be "peak-normalized," which amplifies or attenuates the level of the file to force the highest instantaneous peak to reach 0 dBFS. *This is a very poor way to set the levels of different audio files on a playout system.* Absolute peak levels have nothing to do with loudness, so peak-normalized files are likely to have widely varying loudness levels depending on the typical peak-to-average ratio of the audio in the file. Because of over-use of peak limiting in today's CD mastering (which has the unfortunate side-effect of sucking the life and punch out of music), the average level of a CD produced in the '80s can be as much as 15 dB lower than the average level of a CD produced today. When a playout system segues two such disparate peak-normalized files, this can cause audible loudness inconsistencies in your broadcast/netcast while OPTIMOD-PCn's AGC section corrects the loudness. OPTIMOD-PCn's AGC section uses window-gating technology to minimize the audibility of such gain riding. Even so, 15 dB of level correction can take several seconds and is determined by OPTIMOD-PCn's AGC release time setting.

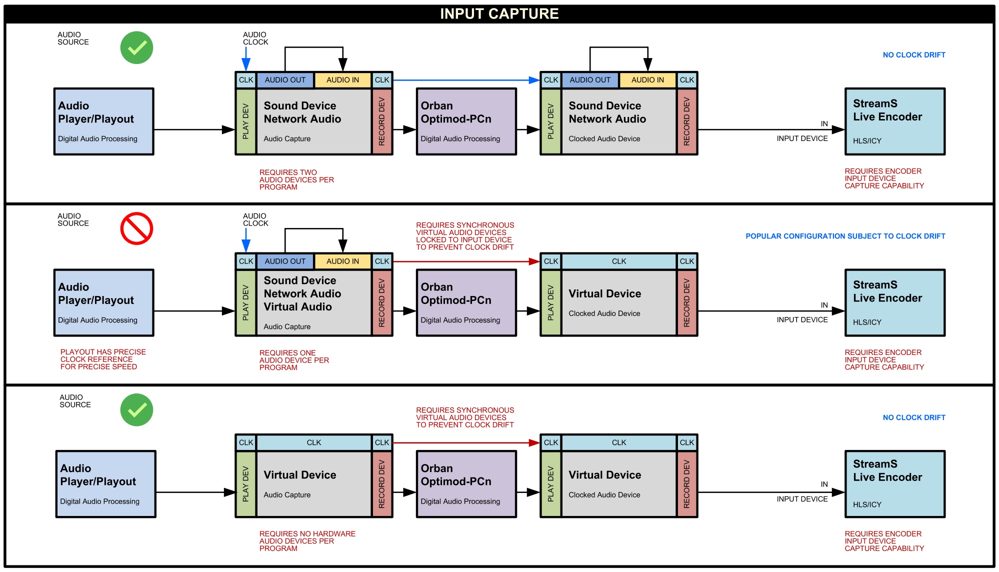
It is far better to normalize levels in a playout system by making the *average* levels of all elements identical, which means that they would all peak at the same level when observed with a VU meter. The average output level of the playout system should be set to about −18dBFS to keep OPTIMOD-PCn AGC nominally in the center of its range. This allows OPTIMOD-PCn's AGC to work as unobtrusively as possible. Moreover, if your system includes locally originated speech material, using a microphone processor (like the dbx 286A, Symetrix 528E, or AirTools 6200/2x) will help smooth the transition between live and recorded program segments.

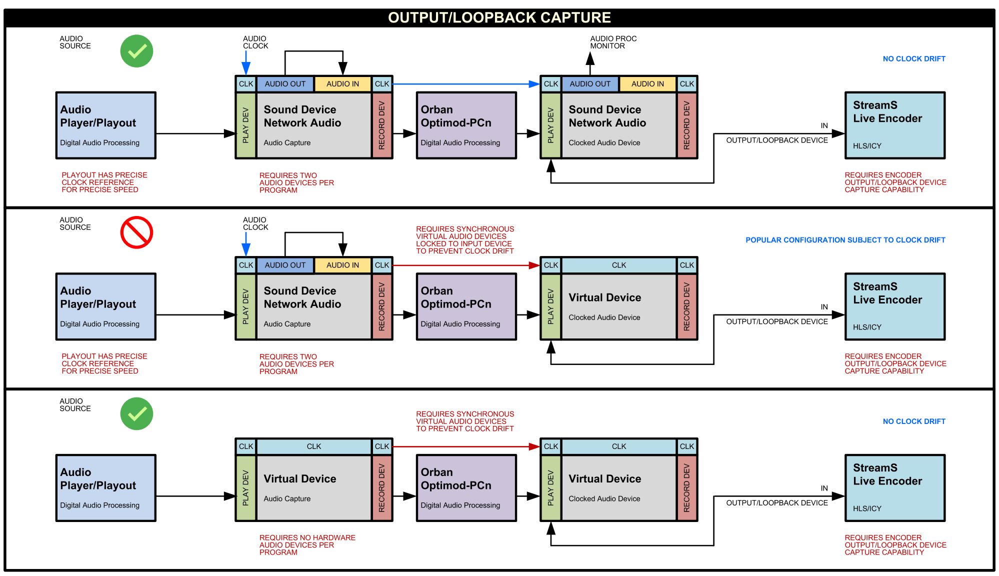
You can also achieve good results by normalizing files so that they all have the same BS.1770 loudness, integrated over the entire file.

Managing Audio Clocking: Block Diagrams

See *Locking Clocks* on page 1-35.

The diagram on page 1-41 shows typical correct (labeled ☑) and incorrect (labeled 🛇) setups with an internal audio source. The diagram on page 1-43 shows correct and incorrect setups with an external audio source. These diagrams can be magnified as needed.





ALL AUDIO SOURCES AND DEVICES MUST REFERENCE THE SAME AUDIO CLOCK.

This includes ALL Virtual Devices. Not all Virtual Devices use the same Audio Clock, cannot be externally referenced, and should be avoided..
Failure to use the same reference clock, can result in audio glitches, dropouts, and audio and metadata time drift.
"The one with two clocks knoweth not the time."

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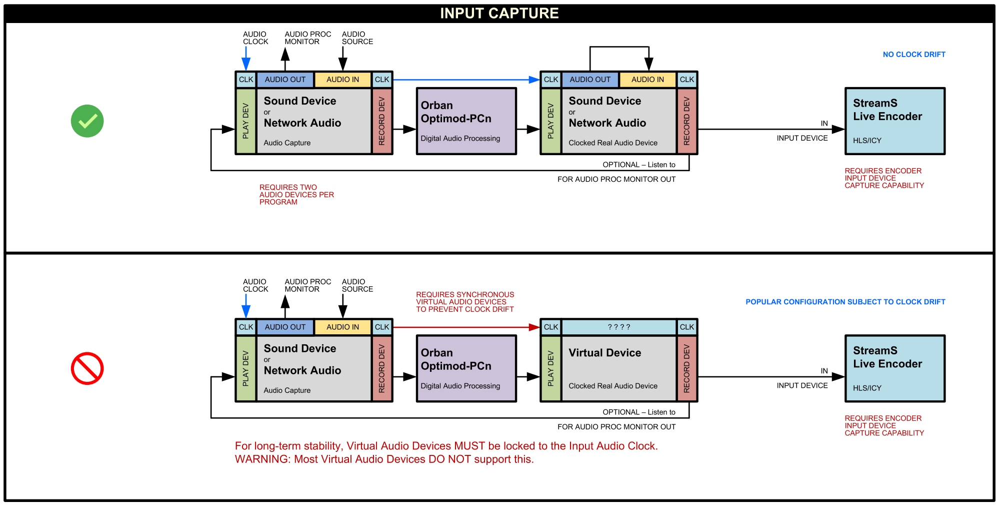
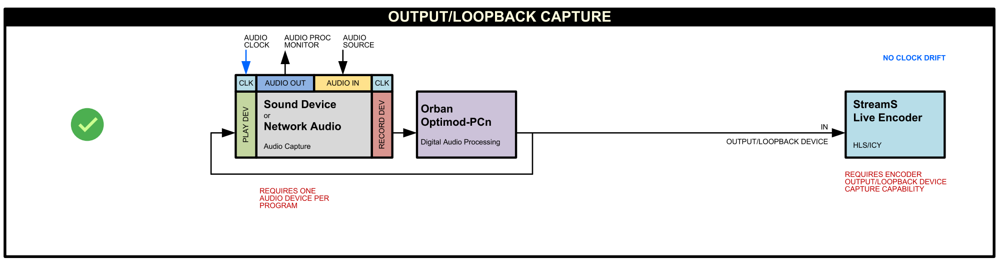
Greg Ogonowski 2019-07

STREAMS™

Streaming WDM Audio Paths – LIVE Internal Audio Sources

Live Streaming Audio Processing and Encoding





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STREAMS ™

Streaming WDM Audio Paths – Live External Audio Sources

Live Streaming Audio Processing and Encoding

Using Lossy Data Reduction in the Audio Chain before OPTIMOD-PCn

Many broadcasters and netcasters are now using lossy data reduction algorithms like MPEG-1 Layer 2 or Layer 3 to increase the storage time of digital playback media. In addition, source material is often supplied through a lossy data reduction algorithm, whether from satellite or over landlines. Sometimes, several encode/decode cycles will be cascaded before the material is finally presented to OPTIMOD-PCn's input.

All such algorithms operate by increasing the quantization noise in discrete frequency bands. If not psychoacoustically masked by the program material, this noise may be perceived as distortion, "gurgling," phasiness, or other interference. Psychoacoustic calculations are used to ensure that the added noise is masked by the desired program material and not heard. Cascading several stages of such processing can raise the added quantization noise above the threshold of masking, such that it is heard. In addition, there is at least one other mechanism that can cause the noise to become audible at the radio. OPTIMOD-PCn's multiband limiter performs "automatic equalization" that can significantly change the frequency balance of the program. This can cause noise that would otherwise have been masked to become unmasked because the psychoacoustic masking conditions under which the masking thresholds were originally computed were changed.

Accordingly, if you use lossy data reduction in the studio, you should use the highest data rate possible. This maximizes the headroom between the added noise and the threshold where it will be heard. In addition, you should minimize the number of encode and decode cycles, because each cycle moves the added noise closer to the threshold where the added noise is heard.

Warranty, User Feedback

User Feedback

We are very interested in your comments about this product. We will carefully review your suggestions for improvements to either the product or the manual. Please email us at info@indexcom.com.

LIMITED WARRANTY

Rorb Inc. warrants Optimod-PCn software and its associated security key (if used) against defects in material or workmanship for a period of one year from the date of original purchase for use, and agrees to repair or, at our option, replace any defective item without charge for either parts or labor.

Software upgrades and technical support expire one year from your date of purchase unless you have purchased a maintenance contract, either when you purchased the OPTIMOD-PCn software or any time before your original support period *expires. After this date, support time extensions are no longer available for purchase, and you must purchase a new version of the software if you wish to upgrade.*

If you purchased Optimod-PCn pre-installed by Modulation Index LLC in a computer, please refer to the separate Modulation Index warranty covering the computer hardware.

IMPORTANT: This warranty does not cover damage resulting from accident, misuse or abuse, lack of reasonable care, loss of parts, or attempts to use this product in computers not having the supported CPU and operating systems (see *Hardware and Software Requirements* starting on page 1-3). No responsibility is assumed for any special, incidental, or consequential damages. However, the limitation of any right or remedy shall not be effective where such is prohibited or restricted by law.

No other warranty, written or oral, is authorized.

This warranty gives you specific legal rights and you may have other rights that vary from state to state. Some states do not allow the exclusion of limitations of incidental or consequential damages or limitations on how long an implied warranty lasts, so the above exclusions and limitations may not apply to you.

Section 2

Installation

Hardware Installation

These Hardware Installation instructions apply to computers pre-configured with 1600 PC software.

Allow about 2 hours for installation.

Installation consists of: (1) unpacking and inspecting the 1600, (2) mounting the 1600 in a rack, (3) connecting inputs, outputs, Ethernet, and power.

DO NOT connect power to the unit yet!

1. **Unpack and inspect.**

A) If you note obvious physical damage, contact the carrier immediately to make a damage claim. Packed with the 1600 are:

Quantity	Item
1	Operating Manual and software on CD
2	Line Cords (domestic, European)
4	Rack-mounting screws, 10-32 x ¾—with washers, #10
1	PC Remote Software CD

B) Save all packing materials! If you should ever have to ship the 1600 (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.

2. **Mount the host computer in a rack.**

There should be a good ground connection between the rack and the host computer chassis — check this with an ohmmeter to verify that the resistance is less than 0.5Ω.

When running the maximum number of OPTIMOD-PCn Processors, the computer's CPU can dissipate more than 100 watts, and you must account for this in your installation. Mounting the unit over large heat-producing devices (such as a vacuum-tube power amplifier) may shorten component life and is not recommended. Ambient temperature should not exceed 45°C (113°F) when equipment is powered.

Equipment life will be extended if the unit is mounted away from sources of vibration, such as large blowers and is operated as cool as possible.

3. **Install the appropriate power cord(s).**

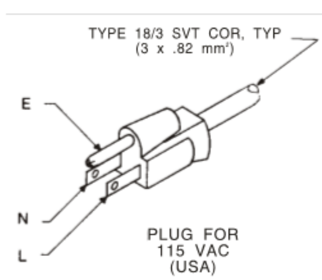
Depending on how they are ordered, some host computers are equipped with dual-redundant power supplies with automatic failover. Each power supply has a dedicated connection to the AC line. For maximum reliability, connect the two supplies to two AC circuits.

AC power passes through an IEC-standard mains connector and an RF filter designed to meet the standards of all international safety authorities.

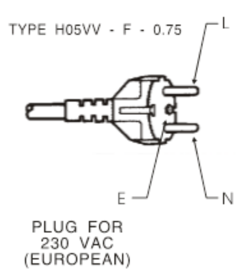
The power cord is terminated in a "U-ground" plug (USA standard), or CEE7 / 7 plug (Continental Europe), as appropriate to your 1600's Model Number. The green/yellow wire is connected directly to the 1600 chassis.

If you need to change the plug to meet your country's standard and you are qualified to do so, see Figure 2-1. Otherwise, purchase new mains cords with the correct line plug attached.

The host computer requires a significant amount of time to boot up, so a brief AC line glitch that causes the computer to restart will cause a significant interruption in audio service and will shut down Windows abnormally. Therefore, **we strong recommend driving the computer from an uninterruptable power supply**. To achieve maximum reliability, connect each power supply to a separate UPS. In addition, a UPS should protect the computer from line surges and minimize the likelihood of damage caused by them.



CONDUCTOR		WIRE COLOR	
		NORMAL	ALT
L	LINE	BROWN	BLACK
N	NEUTRAL	BLUE	WHITE
E	EARTH GND	GREEN-YELLOW	GREEN



CONDUCTOR		WIRE COLOR
L	LINE	BROWN
N	NEUTRAL	BLUE
E	EARTH GND	GREEN-YELLOW

Figure 2-1: AC Line Cord Wire Standard

4. Connect inputs and outputs.

Do this according to the manufacturer's instructions for the Windows input and output devices you will be using. See step 10 on page 2-8.

5. Connect to a network.

The host computer has two Ethernet ports, one dedicated to control and one to audio-over-IP transport, such as Ravenna and Dante. The corresponding audio-over-IP driver software must be installed in the computer. To achieve highest capacity and lowest latency, the audio network should be dedicated only to audio and should not be shared with other data traffic.

We recommend that you control the 1600 via your non-audio LAN from a remote computer running our 1600PC software. This allows the 1600 hardware to run a maximum number of audio Processors and facilitates locating the 1600 in a machine room.

Because the 1600 runs the Windows operating system, you must assign it a static IP address, subnet mask, and gateway using standard Windows procedures. (We assume that you already know how to do this.) Using a static IP allows you to connect reliably to the 1600 from remote computers running our 1600PC remote control software.

If the CPU is not fully loaded, you may also control the 1600 from an instance of 1600PC software running on the 1600 hardware. In this case, the 1600PC software communicates with the 1600's audio processing via a "localhost" connection, which is set up in the 1600 as IP address 127.0.0.1.

Procedures and instructions for connecting to a PC are subject to development and change. We advise you to download the latest version of this manual in pdf format from
https://www.indexcom.com/support/Archive/Orban/1600/Documentation.[1]

You can use Adobe's .pdf reader application to open and read this file. If you do not have the .pdf reader, it is available for free download from www.adobe.com.

Software Installation and Setup

1. Defeat Windows Fast Startup (Windows 8 and higher).

To achieve most reliable startup, OPTIMOD-PCn Service is set for DELAYED START in Windows and starts after a 60-second delay. Windows 8 introduced Fast Startup, which overrides the delayed start during a power-on Windows boot. To ensure that OPTIMOD-PCn starts reliably on power-up, you must defeat Windows FAST STARTUP:

A) Navigate to CONTROL PANEL > ALL CONTROL PANEL ITEMS > POWER OPTIONS.

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[1] For login credentials, contact: support@indexcom.com

B) Click CHOOSE WHAT THE POWER BUTTONS DO on the left side of the screen.

C) Click CHANGE SETTINGS THAT ARE CURRENTLY UNAVAILABLE.

D) Under SHUTDOWN SETTINGS, *uncheck* TURN ON FAST STARTUP (RECOMMENDED).

2. **Install 1600PC software on the computer that will control the 1600.**

   1600PC software will be installed when you install the Service. However, in most cases, it is preferable to control the 1600 from another computer because this will maximize the amount of CPU power dedicated to processing audio.

   To control the 1600 from another computer, you must install 1600PC software on that computer. Because the 1600PC application is graphics-intensive but is otherwise requires only a small amount of CPU power, almost any modern computer will work if it has competent graphics support.

   > 1600PC software is not copy-protected; you may install it on as many computers as you wish. Remember that the 1600PC application does not do audio processing; it only controls OPTIMOD-PCn Service(s) running on local or remote computers. The Service does the audio processing and is copy-protected via a hardware key or via a CodeMeter software license.

   A) From https://www.indexcom.com/support/Archive/Orban, download the installer program, Orban-Optimod-PCn_1600-1601-1602_x.x.x.x.exe, where "x.x.x.x" represents the software version. (For example, for version 1.0 software, this would be 1.0.0.0.) Downloading ensures that you get the latest software version. For login credentials, contact: support@indexcom.com.

   B) Run Orban-Optimod-PCn_1600-1601-1602_x.x.x.x.exe on its host computer by right-clicking its icon and selecting RUN AS ADMINISTRATOR. The installer will start up. Follow the instructions it provides on each screen. In the SELECT COMPONENTS screen, choose OPTIMOD APPLICATION FOR REMOTE ACCESS.

   This program installs the necessary files and adds an Orban/Optimod 1600 folder to your computer's Start Menu. This folder contains shortcuts to the PC Remote application and to the documentation. If you accepted the option during installation, there is also a shortcut to the PC Remote application on

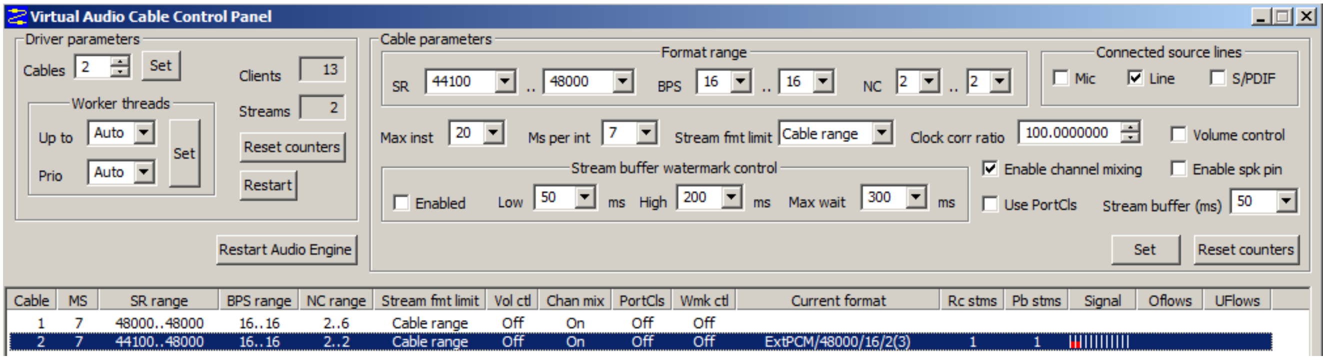


Figure 2-2: Virtual Audio Cable Setup

your desktop.

3. **If needed, install Virtual Audio Cable on the server computer.**

   If you need to connect audio from one application to another (such as the output of a playout system to the input of OPTIMOD-PCn), we recommend using Virtual Audio Cable. It is available for purchase from its vendor:

   http://software.muzychenko.net/eng/vac.htm

   VAC consists of an audio driver that creates Windows Recording and Playback sound devices, and a control application. You must set up one "cable" for every interconnection you require between applications. While it is beyond the scope

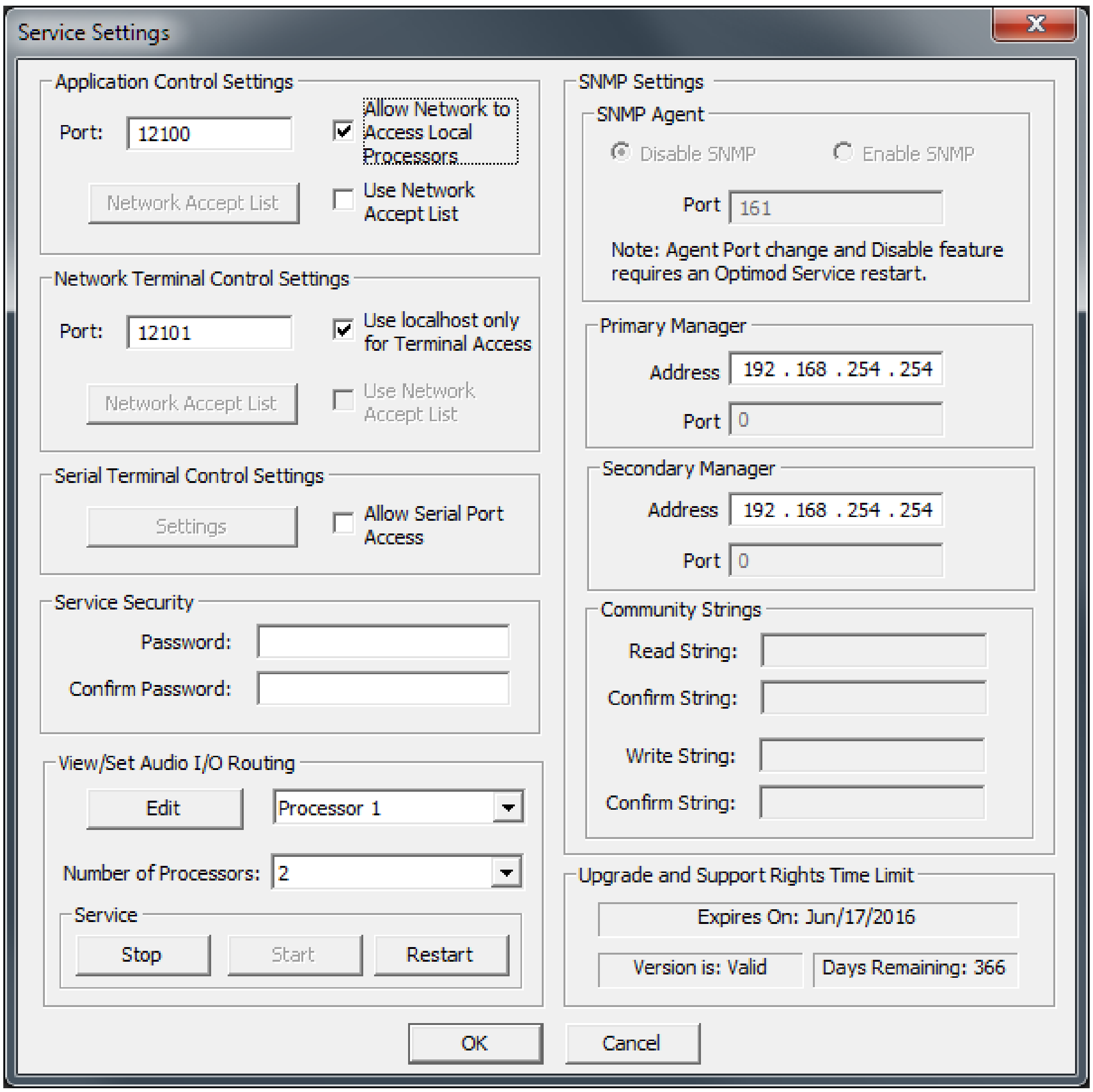


Figure 2-3: Service Settings

of this manual to provide detailed setup instructions, Figure 2-2 on page 2-4 shows a typical setup, where Cable 2 connects the output of a playout system to the input of OPTIMOD-PCn. *Carefully note the usage limitations in* Locking Clocks *on page 1-35*.

4. **Authorize the server computer.**

   - *If your 1600PCn is protected by a hardware key:* Plug the 1600PC copy protection key (supplied with your software) into an available USB port of the computer you will be using to do audio processing. Any computer running the OPTIMOD-PCn Service must be authorized.

     The key does not require a Windows driver.

     If the computer is already running without the key, inserting the key will not automatically start the Service. Do this from the 1600PC TOOLS > SERVICE SETUP window.

   - *If your 1600PCn is protected by CodeMeter software authorization:* The software installer (step 5 below) will install the document *Instructions to create license request file for Optimod PCn.pdf* and offer to open it. Follow these instructions to obtain your authorization.

5. **Install 1600PC software on the computer that will run the Service and perform audio processing.**

   Follow the instructions in step 1 above, except choose OPTIMOD APPLICATION AND SERVICE in the SELECT COMPONENTS screen. Be sure that you run the installer as Administrator.

   During installation, the installer will automatically create an Inbound Rule in the Widows Firewall to permit connection to the `OptimodService3.exe` application. The firewall must permit inbound connections if you wish to control OPTIMOD-PCn remotely.

6. **Set up the Service.**

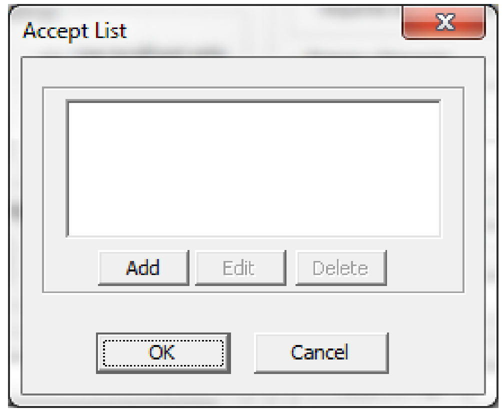
   NOTE: You can only set up the Service via a 1600PC running on the same PC as the Service, and you must have Administrator rights to do so. You cannot use a remote instance of 1600PC for this setup. If you need to set up the Service remotely, use Windows Remote Desktop to control a 1600PC running on the Service computer.

   A) Start up the 1600PC application on the computer running the Service.

   B) Open the SERVICE SETTINGS window from TOOLS > SERVICE SETTINGS.

   C) In the NUMBER OF PROCESSORS drop-down, select the number of independent audio processors you wish to run. This is limited to the number of processors you purchased, which is encoded in your security key or CodeMeter authorization, depending on which version of 1600PCn you have.

      About the Upgrade and Support Rights Time Limit Field: The version of OPTIMOD-PCn software you have purchased never expires, but free upgrades and technical support are available for a limited time. The cutoff



date and the number of Processors you have purchased are encoded in your license key or CodeMeter authorization. The date shown is the last day that free technical support is available for your license, and you can only run versions of the software released before this date. You may purchase an extension any time before this date by contacting info@indexcom.com. After this date, extensions are no longer available for purchase, and you must purchase a new version of the software if you wish to upgrade.

7. Set up application control settings.

These settings can allow 1600PC applications anywhere on your network to connect to the Service.

A) Choose the Port that you will use to control the Service from 1600PC.

The default is 12100.

B) If you wish to allow 1600PC running on other computers to connect to the Service, check Allow Network to Access Local Processors.

By default, this box is not checked. When it is not checked, no Processor in your computer can be accessed through the OPTIMOD-PCn Control Application running on the network.

Before you check this box, be sure that you have assigned a password to each OPTIMOD-PCn Processor in your computer (step 12.D) on page 2-11).

C) If you wish to only allow certain IP addresses to connect to the Service:

a) Check the Use Network Accept list box.

b) Click the Network Accept List button.

c) Click the Add button.

d) Enter host name, domain name, IP address, or subnet address in CIDR format.

Examples:

```
host.domain.com                Single Computer
domain.com                     Entire Domain
123.45.67.8                    Single Computer
123.45.67.8/24                 Entire IP (subnet) Range
123.45.67.8/255.255.255.0      Entire IP (subnet) Range
```

e) If you wish to add another computer, repeat steps (a) through (c).

f) When you are finished adding computers, click OK.

8. Set up Network Terminal Control Settings

The OPTIMOD-PCn Service application hosts a TCP/IP terminal server to allow external control of the OPTIMOD-PCn Processors from either a Telnet/SSH client or a custom third party application. Many controls are accessible and all commands are simple text strings. Upon receiving valid commands, OPTIMOD-PCn will con firm by returning a simple text string status message. By implementing external control this way, multiple OPTIMOD-PCn Processors can be controlled using standards-based network protocols (that are not Microsoft Windows-specific) anywhere that network connectivity is available.

A) The IP address of the terminal server is the same as the address of the computer hosting the Service. The default Port is 12101, but you can change this as desired.

B) If you wish to allow only Localhost (127.0.0.1) access to the terminal server, check USE LOCALHOST ONLY FOR TERMINAL ACCESS.

9. Set up security for the Service

In any but the simplest and most secure installations, we strongly recommend using password protection to limit access to the SERVICE SETTINGS. Enter it in the SERVICE SETTINGS > SERVICE SECURITY / PASSWORD box.

10. Set up audio I/O for each Processor.

Each Processor must have a Windows Recording sound device assigned to its input and a Windows Playback sound device assigned to its output. You may use the same device for the input to more than Processor, but each output must be assigned to a different device. Before you configure OPTIMOD-PCn you must first configure Windows to be consistent with the OPTIMOD-PCn setup you are planning to use. *The sample rates of the input and output devices must be locked together*; see *Locking Clocks* on page 1-35.

A) Set up the Windows Sound Devices you will be using.

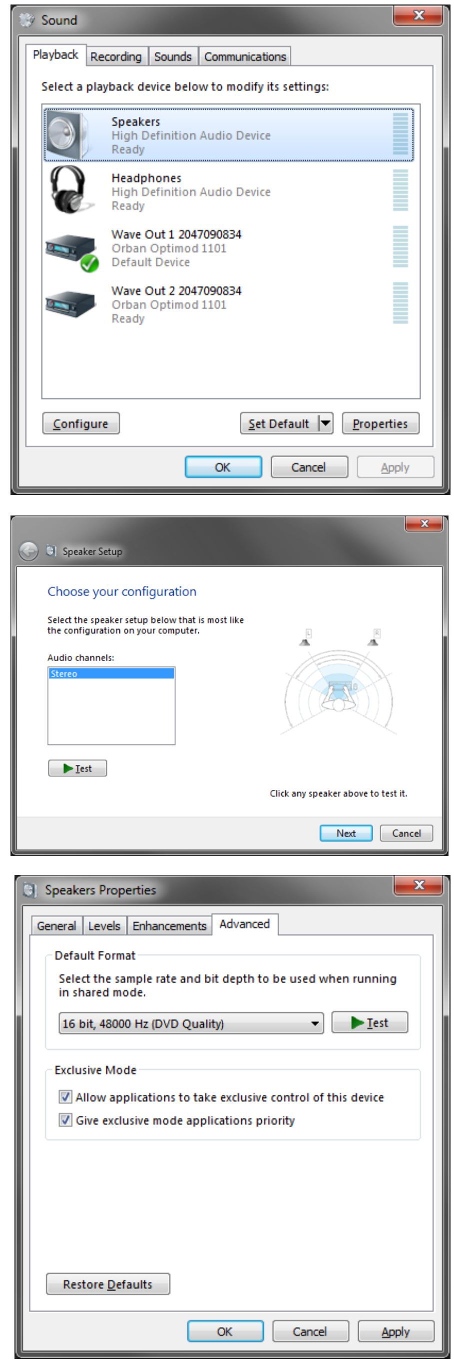


Figure 2-4: Windows Audio I/O Setup

a) Go to Windows CONTROL PANEL> SOUND> RECORDING> PROPERTIES> ADVANCED. For mono or stereo operation, configure each Recording device for configure each Recording device for your choice of 44.1 kHz, 48 kHz, 96 kHz, or 192 lHz sample rate, 16-bit or 24-bit, 2 Channel.

b) Go to Windows CONTROL PANEL> SOUND> PLAYBACK. Depending on whether or not you want to be able to send an stereo→5.1 upmix to a given Playback (output) device, highlight the device, set it for configure each Recording device for the same sample rate and bit depth that you used for the Recording Device (in PROPERTIES > ADVANCED), and 2 Channel or 5.1 Channel (in CONFIGURE).

Many I/O devices support only Stereo / 2 Channel. 5.1 will usually require using the “Speakers” playback device.

The drivers of the Playback devices you choose must support Windows WASAPI. Most do, as Microsoft introduced and standardized this model starting with Windows Vista.

For both Recording and Playback devices, check ALLOW APPLICATIONS TO TAKE EXCLUSIVE CONTROL OF THIS DEVICE and GIVE EXCLUSIVE MODE APPLICATIONS PRIORITY.

c) If the ENHANCEMENTS tab is present for a given playback device, click it and check DISABLE ALL ENHANCEMENTS.

B) Set up OPTIMOD-PCn’s inputs and outputs.

a) In SERVICE SETTINGS > VIEW/SET AUDIO I/O ROUTING, choose the Processor you wish to set up.

b) Specify the input device from the drop-down menu. This must be a device that you configured in step A)a) on page 2-8.

c) Choose the processing mode and output channel mode:

[2.0→2.0 STEREO]
[2.0→5.1 UPMIX],
[1.0 MONO]
[5.1 SURROUND (6-IN, 6-OUT)]
[5.1 SURROUND + 1 DOWNMIX (6-IN, 8-OUT)]

These choose mono processing (using 2-channel I/O), stereo processing with 2-channel I/O (no upmixing), stereo processing with 2-channel input and 5.1-channel output (to support an upmixed output), 5.1 surround processing, and 5.1 processing with additional 2.0 downmix processor with its own dedicated loudness controller. 5.1 inputs are only available if you have purchased one or more surround instances of OPTIMOD-PCn.

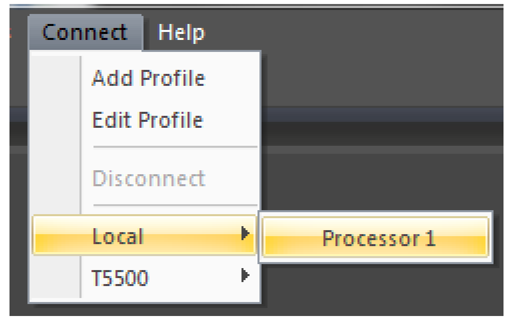
The specified Windows Recording and Playback devices must be set up to match the processing mode [step (A) on page 2-8] or OPTIMOD-PCn will issue an error message when you click OK.

Because many Windows sound devices do not support a true mono mode, OPTIMOD-PCn’s Mono mode uses stereo (2.0) input and output devices. The main benefit of Mono processing is that it cuts CPU usage by approximately 50% compared to Stereo processing.

d) Click OK.

C) Repeat step (B) for each Processor you are setting up.

D) Click OK to dismiss the Service Settings dialog box. *This will restart the Service and briefly mute the audio.*



11. Local installations: Connecting to an OPTIMOD-PCn Processor.

If 1600PCn software is running on the same computer that runs the Service, connect to Processors via Connect > Local.

12. Set up passwords and names for Processors in your computer. (optional)

Before you perform this and subsequent steps, please read *Overview of an OPTIMOD-PCn Installation* starting on page 1-30. OPTIMOD-PCn is versatile, so it is important to understand the information in that section.

Each Processor in OPTIMOD-PCn 1600 software appears to the 1600PC remote application as a separate, individually addressable audio processor. The number of instances you have purchased determines the maximum number of instances permitted on your system. However, turning on CPU-intensive parts of the OPTIMOD-PCn 1600 processing, like the MX limiter or Optimix upmixer, may limit the number of available instances to fewer than the maximum because all of the CPU power has been used up.

> If the CPU is overloaded, you will hear glitches and stuttering in the audio and, in Windows Task Manager on the 1600, you will probably see one or more cores of the CPU sitting at close to 100% utilization. Each Processor must run almost entirely in one core, although several Processors can share one core if that core has enough resources to run them smoothly. Note that two Hyperthreaded cores share the CPU resources of a given physical core, so the effective clock speed of a Hyperthreaded core is slightly more than half that of a physical core. See Table 1-1 on page 1-5 and the associated discussion.

Logically, instances are almost independent of each other. 1600PC remote software connects to one instance at a time. Each instance can run its own processing presets regardless of what presets other instances are running.

A) On the computer running the Service, start the OPTIMOD-PCn control application, `1600PC.exe`, by clicking its icon (“agent”) in your computer’s system tray, or from Start > Programs > Orban > Optimod PCn 1600.

> Of course, you may create Windows shortcuts to 1600PC using the normal Windows mechanisms.

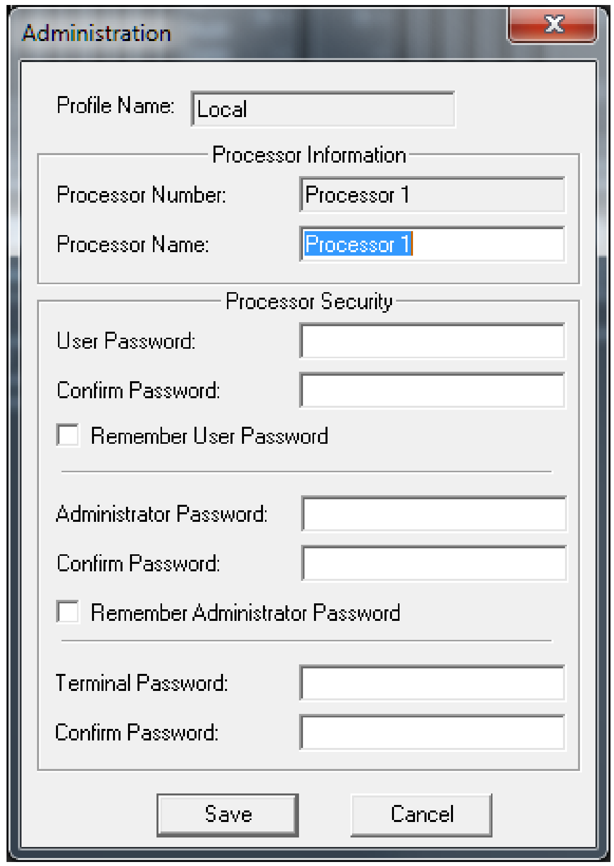
Newly installed Processors have no passwords. You do not need to assign a password to a Processor unless you want that Processor to be accessible to a network connected to your computer.

> Once you have assigned passwords, these passwords can be changed either locally or on the network by anyone with the Administrator pass-

word for that instance, but no one else. Do not lose the Administrator password you assign in the steps below.

B) Connect to the Local Processor to which you are assigning a name and password.

C) From the Tools menu bar, choose "Administration."



Initially, OPTIMOD-PCn software identifies a Processor as Processor[x], where the numbering starts at 1 and increases in steps of 1 for each Processor. You can give the Processor an easily remembered name ("alias") by filling in the PROCESSOR NAME field. OPTIMOD-PCn software can then identify the Processor by this name from anywhere on your network.

If you had previously set up 1600PC on a remote computer to include the Profile for this service computer [step (15.A) on page 2-13], the Profile on the remote computer will not update automatically to show a Processor's revised Processor Name. To make the revised name appear, Delete the Profile on the remote computer and Add it again.

D) Assign a User Password to the Processor by filling in the USER PASSWORD and CONFIRM PASSWORD fields identically. This password allows you to connect to a Processor via 1600PC.

E) Assign a Terminal Password to the Processor by filling in the TERMINAL PASSWORD and CONFIRM PASSWORD fields identically. This password allows you to connect to a Processor via OPTIMOD-PCn TCP/IP terminal server to allow external control of the OPTIMOD-PCn Processors from either a Telnet/SSH client or a custom third party application. See *OPTIMOD 1600PCn Control* API on page 5-1.

F) Assign an Administrator Password to the Processor by filling in the ADMINISTRATOR PASSWORD and CONFIRM PASSWORD fields identically.

To preserve security, do not make the User and Administrator passwords identical.

Be sure to write down and remember the Administrator Password because you must have it to change the Processor's User Password or the Processor Name in the future. You may check REMEMBER ADMINISTRATOR PASSWORD, but be aware that this will allow *anyone* with access to the Service computer to change the User *and* Administrator passwords for this Processor. Used maliciously, this privilege could lock you out of the Processor, requiring an inconvenient uninstallation, reinstallation and reconfiguration of the Service.

G) Click SAVE to confirm your entries.

H) Repeat steps (B) through (G) for each new Processor installed in your local computer.

When you re-enter the CONNECT menu, you will now see the Processors listed by the names you have assigned to them, not by their Processor Numbers.

13. Edit the local OPTIMOD-PCn server's network accessibility, port numbers, and Service Security Password. (optional)

A) If the OPTIMOD-PCn application is currently connected to a Processor, disconnect it by choosing DISCONNECT from the CONNECT menu.

B) Navigate to TOOLS > SERVICE SETUP.

C) Enter the Service Security Password that you originally set in step 9 on page 2-8.

The Service Setup Window appears.

D) Edit parameters in this window as desired. Then click OK. (See step 7 on page 2-7.)

14. Edit a given Processor's name and/or passwords. (optional)

A) Connect to the local Processor to which you are assigning a name and password.

B) From the Tools menu bar, choose "Administration."

The Enter Administrator Password window appears. Enter the Processor's Administrator password and hit *Enter*.

> If you did not enter an Administrator Password in step (12.E), you will not see the "Enter Administrator Password" dialog box and the Processor and Security Administration window will appear immediately.

C) The Processor and Security Administration window opens. Edit the fields as required. (See step 11 on page 2-10.)

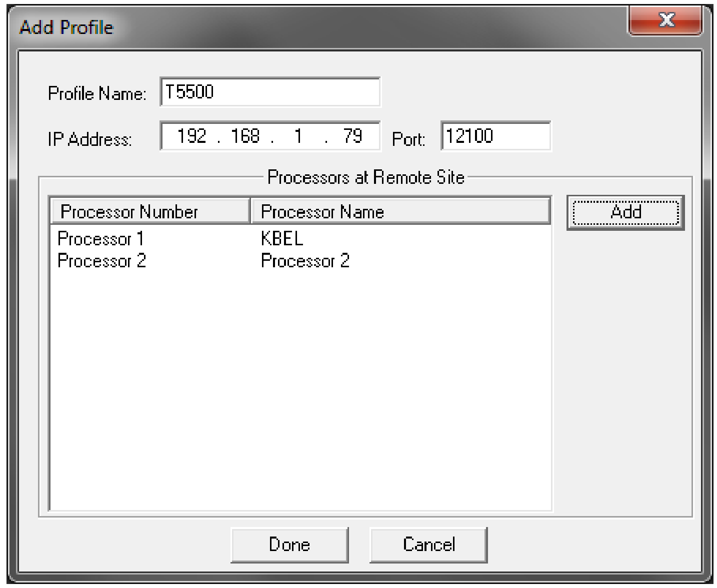
D) Click "Save" to confirm your changes.

15. Make Processors visible on remote computers (i.e. those not hosting the Service). (optional)

To aid security, all OPTIMOD-PCn Processors are initially hidden from 1600PC applications running on computers other than the computer hosting the Service for those Processors. On each remote-control computer, you must explicitly "add" each Service computer so that its Processors appear on the Connection list of the 1600PC application running on the remote-control computer.

To make a Processor controllable from a remote-control 1600PC, you must know three things: (1) the IP address of the Service computer implementing the Processor, (2) the port that was assigned to the Service (default is 12100), and (3) the User or Administrator password of each Processor.

To add Processors, you must be able to connect to the Orban server software on the Service computer running those Processors:



- The Service computer must be online and connected to the network.
- The Orban server software on the Service computer must be running, which means that the security key must be plugged into a USB socket or CodeMeter authorization software must be running on the Service computer.
- The Windows firewall on the Service computer must allow OptimodService3.exe access to the network. (See *Using Windows Firewall with* OPTIMOD-PCn on page 2-37.)
- The "Allow Network to Access Local Instances" box in the Service computer's SERVICE SETUP [step (7.B) on page 2-7] must be selected.

Computers not actually doing audio processing should not have the OPTIMOD-PCn Service installed on them; 1600PC does not require (or use) the Service to connect to remote Service computers. However, in a network connected to more than one Service computer it is possible to make other Service computers visible to 1600PC running on one of them. To do this, Add the Profiles of the additional Service computers one at a time using the instructions in step (A) below. In this case, 1600PC can still only control the Service Settings for the computer on which it is installed; the Service Settings of other Service computers on the network can only be accessed via their local 1600PC applications.

A) Add a Service computer to the list of available computers containing OPTIMOD-PCn instances:

a) Select CONNECT>ADD PROFILE.

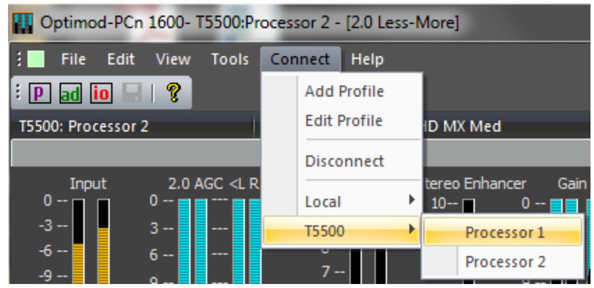
The ADD PROFILE window appears.

b) Enter the Profile Name, IP Address, and Port of the Service computer.

The Profile Name can be any name you wish to use—for example, "Remote." The profile name is known only to your local computer. The network identifies a given Service computer by its IP address and port, not its profile name. The profile name is merely a convenient alias that you use to help identify a Service computer hosting OPTIMOD-PCn Processors without your having to memorize the computer's IP address.

c) Click Add.

If 1600PC can connect to the Service computer, the Processor Names of all Processors running on that Service computer will automatically appear in the PROCESSORS AT REMOTE SITE list. 1600PC will add them to the Connect menu and the Connection List window. You can open the Connection List window by selecting it from the VIEW menu.



B) Add remote computer Profiles and the instances in each Profiled computer as desired, by repeating step (A) for each remote computer.

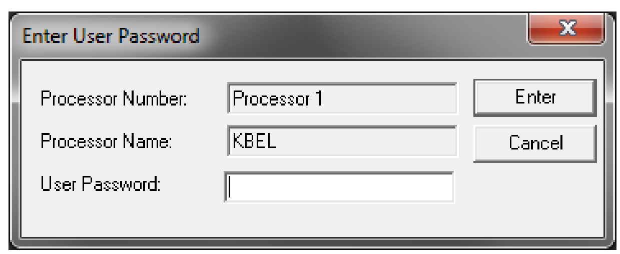
16. Connect to a remote Processor. (optional)

You can now connect to any Processor that you added in step 11 on page 2-10 if this Processor's Service is running and connected to the network.

A) Click the CONNECT menu. A drop-down menu appears containing a list of all remote computers. Drag your mouse down to the desired computer to reveal a submenu containing all Processors within it that have been added. Select the desired Processor to connect to it.

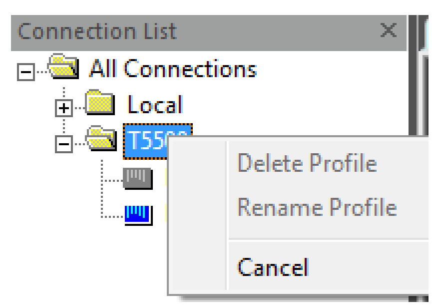
B) The "Enter User Password" dialog box appears. Enter the password and click ENTER. If you wish to bypass this dialog box automatically in the future, check the REMEMBER USER PASSWORD box in the ADMINISTRATION screen on your local computer [step (12.C) on page 2-11]. (Only Administrators of your local Machine can restore this dialog box once you have specified that it is to be bypassed.)

After you click ENTER, the Control application will display the state of the Processor to which you just connected, and you can recall presets, adjust input/output levels, edit and save presets, etc.



17. Delete a remote computer's profile. (optional)

If you no longer wish to have a particular remote computer appear in the list of available computers, you can delete it from the list. Additionally, Deleting a profile and then Adding it again will refresh the Processor Names if you changed these on the Service Computer.



A) Disconnect from any Processors using the CONNECT > DISCONNECT menu item.

B) Open the CONNECTION LIST window using VIEW > CONNECTION LIST.

C) Right-click the remote computer you wish to remove and select DELETE PROFILE.

The remote computer's profile (and the profiles of all of its corresponding Processors) are removed from your local computer. This action does not affect any other computer on the network.

You can restore the remote computer's profile by following the instructions in step (15.A) on page 2-12.

Setup: The OPTIMOD-PCn Control Application

Once you have connected the Optimod-PCn Control application to an OPTIMOD-PCn Processor, the software allows you to control the Processor as if it were a dedicated hardware processor. The Control application displays gain reduction meters, as well as controls that allow you to edit the sound of the factory presets to your liking. Section 3 of this manual explains these sound editing controls in detail.

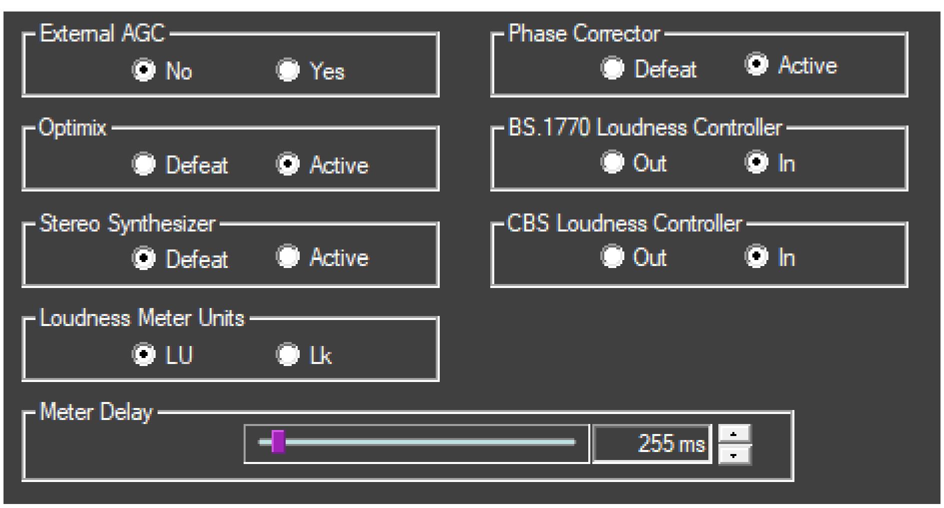
1. Select the Processor you are setting up.

A) From the Connect menu on the Control application, select the Processor you are setting up by connecting to it.

If you wish, you can edit the factory default Input/Output ("IO") setup. You also have the option to OPEN SETUP or SAVE SETUP AS in the Control application's FILE menu. Therefore:

- You can save your current setup to a file and then apply it to another Processor.
- You can load a previously saved setup into the Processor that is currently selected

2. Activate or defeat processing blocks as needed.



OPTIMOD-PCn allows you to deactivate processing blocks that you do not need. This can reduce input/output delay and/or reduce CPU usage.

A) From the TOOLS menu, open the I/O Mixer and click the UTILITY tab.

B) Set each switch as appropriate.

- OPTIMIX: Defeating Optimix removes about 212 ms of delay and reduces CPU usage substantially because the Optimix upmixer does not operate, nor does the center-channel MX limiter when the MX limiter is activated in a given processing preset. If this switch is ACTIVE, these elements operate even if Optimix is bypassed in the active processing preset. This prevents glitches when activating or defeating Optimix within the processing preset.

  Optimix is defeated automatically if the processor's output is configured as STEREO in step 10.B)c) on page 2-9. The Optimix tab will not appear.

- STEREO SYNTHESIZER: The stereo synthesizer is not a compute-intensive process, but it introduces about 187 ms of look-ahead delay to allow its automatic mode sensing to work (by sensing silence in the right channel). Defeating the stereo synthesizer removes this delay.

- PHASE CORRECTOR: The phase corrector is a moderately CPU-intensive process that introduces about 170 ms of delay. Unlike Optimix, it does not operate in the background when it is defeated in the active processing preset. Therefore, the main benefit to defeating it in the System is to remove its delay.

- BS.1770 LOUDNESS CONTROLLER: This can defeat the BS.1770 safety limiter. The BS.1770 meter runs regardless of the setting of this control.

- CBS LOUDNESS CONTROLLER: This defeats the CBS loudness controller and loudness meter. The CBS loudness controller/meter relies on a large filterbank, so it uses a significant amount of CPU, and defeating it can sometimes help achieve smooth operation on slower CPUs.

- LOUDNESS METER UNITS: This determines the label of the BS.1770 loudness meter and only affects the graphics. LkFS is typically used in the United States and LUFS in Europe.

C) Set the EXTERNAL AGC mode as appropriate for your installation.

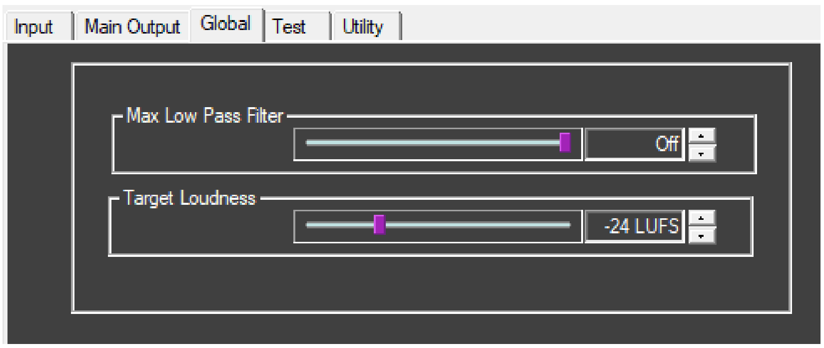
[Yes] or [No]

This control turns OPTIMOD-PCn's internal AGC (Automatic Gain Control) on or off.

In radio applications, it is common to have an external AGC at the studio side of a studio-to-transmitter link to protect the link from overload. Most of the processing structures in OPTIMOD-PCn control level with a preliminary AGC internal to OPTIMOD-PCn. If you are using an external AGC device (such as an Orban 6300 multipurpose audio processor or another OPTIMOD-PCn) in front of OPTIMOD-PCn, set OPTIMOD-PCn's internal AGC to DISABLED. This is to ensure that the internal and external AGCs do not "fight" each other and that they do not simultaneously increase gain (resulting in increased noise).

Temporarily set AGC to ENABLED so that the Analog and Digital Input reference level alignment steps (below) will work correctly. After you have finished with these steps, set the AGC parameter appropriately for your installation.

3. Configure global audio processing parameters.



A) From the TOOLS menu, open the I/O Mixer and click the GLOBAL tab

B) Set the MAX LOWPASS FILTER cutoff frequency.

[10.0 kHz] to [20.0 kHz]

There are two sets of controls, one for SURROUND PROCESSING and one for DOWNMIX PROCESSING. SURROUND refers to the main audio processor, which includes the Optimix upmixer if configured for 5.1 output. DOWNMIX refers to the downmix processor, which always emits a stereo downmix of the audio regardless of whether Optimix is active or bypassed.

The DOWNMIX processor is not supported in v0.9 software. Only the SURROUND PROCESSING controls work.

You can set OPTIMOD-PCn's audio bandwidth in two places:

(1) in the I/O Mixer's GLOBAL tab, and (2) in the main window's EQ tab. (If the EQ tab is not visible, click the "ad" button on the toolbar.) OPTIMOD-PCn's bandwidth is always the *lowest* of these settings. The frequency in GLOBAL is a technical parameter that determines the *highest* bandwidth available. The installing engineer should set it to be appropriate for the sample rate of the digital system that OPTIMOD-PCn is driving. For example, if OPTIMOD-PCn is driving a system with a 32 kHz sample rate, set the MAX LOWPASS FILTER cutoff frequency to 15.0 kHz. That way, a setting of 20 kHz in the EQ tab will not cause excessive bandwidth and aliasing because OPTIMOD-PCn will automatically override it with the MAX LOWPASS FILTER cutoff frequency setting.

NOTE THAT THE LP FILTER ON THE EQ TAB IS PART OF THE ACTIVE PRESET, LIKE ANY OTHER EQUALIZATION CONTROL IN THE PRESET. IF YOU RECALL A DIFFERENT PRESET, THE LOWPASS FILTER CUTOFF FREQUENCY CONTROL IN THE NEW PRESET WILL NOW DETERMINE THE SYSTEM BANDWIDTH (UNLESS, OF COURSE, THE Lowpass Filter

CUTOFF FREQUENCY SETTING IN THE NEW PRESET IS HIGHER THAN THE Max Low-pass Filter CUTOFF FREQUENCY PARAMETER IN THE Global PAGE).

4. Set Digital Output properties.

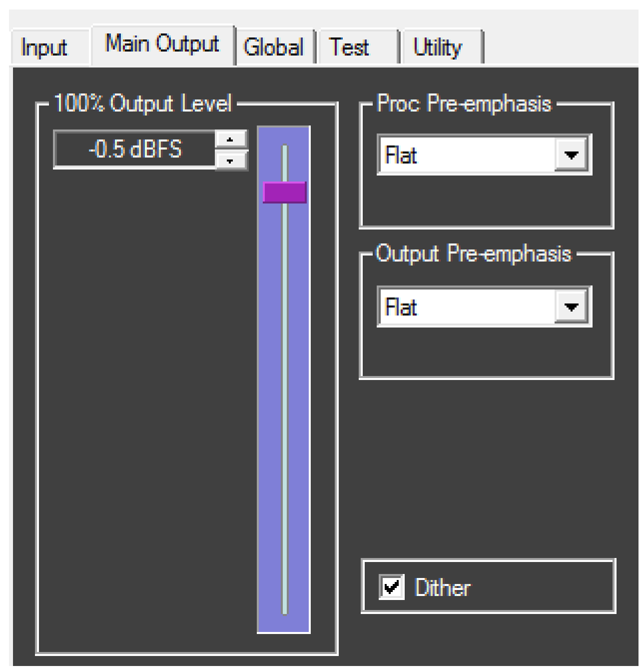
A) Navigate to I/O > STEREO OUTPUT.

B) Adjust DITHER to on or off, as desired.

[in] or [out]

OPTIMOD-PCn can add "first-order high-pass" dither before any truncation of the output word. The amount of dither automatically tracks the setting of the WORD LENGTH control.

OPTIMOD-PCn's dither is first-order noise shaped dither that adds less noise in the midrange than white PDF dither. However, unlike extreme noise shaping, it adds a maximum of 3dB of excess total noise power when compared to white PDF dither. It is therefore a good compromise between white PDF dither and extreme noise shaping.



In many cases, you will not need to add dither because the source material has already been correctly dithered and more than enough input dither is passed through to OPTIMOD-PC's output. However, particularly if you use the Noise Reduction feature, the processing can sometimes attenuate input dither so that it is insufficient to dither the output correctly. In this case, you should add dither within OPTIMOD-PCn. It is safest to always add dither when operating with 16-bit output.

C) Set output and Target Loudness parameters to achieve the desired target loudness and output headroom.

a) Set the 100% OUTPUT LEVEL control to the amount of the peak headroom you need to match the downstream transmission channel.

In some applications, it is important to leave peak headroom below 0 dBFS to accommodate overshoots in the downstream transmission system. For example, when using a lossy codec like HE-AAC, it is wise to allow at least 1.5 dB of peak headroom to compensate for codec-induced overshoots.

The 100% OUTPUT LEVEL control sets OPTIMOD-PCn's maximum peak output level with respect to 0 dBFS, which allows you to compensate for transmission channels that introduce peak overshoots. *It is effectively a* *peak limiter threshold control and does not change loudness.* Hence, the average output level does not change but the maximum peak output level is constrained to the control's setting[2]. If you have set TARGET LOUDNESS correctly on OPTIMOD-PCn, you can change the output level control freely without causing your loudness to be incorrect with respect to other transmissions.

OPTIMOD-PCn's peak limiter is "true-peak"-aware (per ITU BS.1770), so the peak headroom that you specify will be preserved even after sample rate conversion or D/A conversion.

b) *To adjust loudness, adjust the active TARGET LOUDNESS value.* This control is in the active processing preset in the LESS-MORE tab. If the processing preset's TARGET LOUDNESS = GLOBAL, then OPTIMOD-PCn instead uses the global SURROUND TARGET LOUDNESS value in I/O SETUP > GLOBAL.

For more detail, refer to *About Target Loudness and ITU-R BS.1770* starting on page 1-16 and *Setting Loudness* starting on page 1-20. *Customizing Processing Presets to Achieve Specified Target Loudness* starting on page 2-23 provides information on customizing processing presets to achieve a specified BS.1770 loudness.

Your Optimod's Target Loudness and output level controls are only calibrated correctly if the transmission channel after the Optimod has unity gain. Because gain > 1 can cause clipping and gain < 1 will cause loss of headroom in the transmission channel, non-unity gain after the Optimod is bad engineering practice.

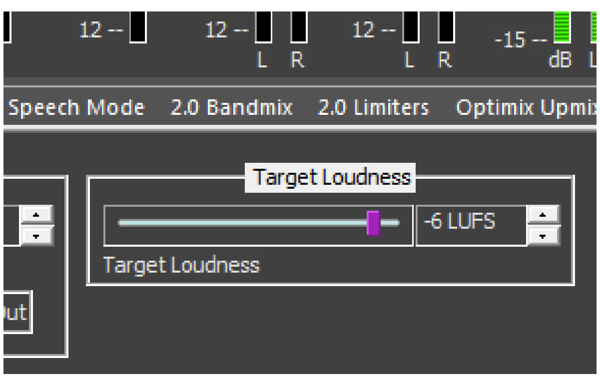


Figure 2-5: Target Loudness in Processing Preset

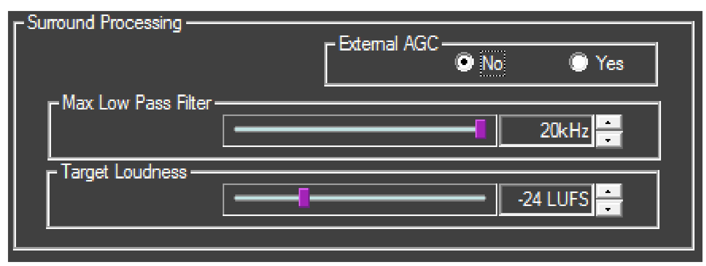


Figure 2-6: Target Loudness in I/O SETUP

Unlike most other Optimods, the LESS-MORE control in a processing preset affects texture while changing loudness as little as possible. Set loud-

[2] If this causes large amounts of gain reduction to occur in the look-ahead limiter, the average level will decrease somewhat. However, this will only happen with pathologically high settings of Target Loudness and will never happen at the settings of TARGET LOUDNESS typically used in broadcasts (–24 per ATSC A/85 or –23 per EBU R128).

ness by adjusting the active Target Loudness value. “TV” presets use the global TARGET LOUDNESS value; other presets usually have local TARGET LOUDNESS values.

For a detailed explanation, refer to Setting Loudness starting on page 1-20. Customizing Processing Presets to Achieve Specified Target Loudness on page 2-23 provides further detail.

D) Set preemphasis curve on which the peak limiting operates.

[Flat], [50 µs], [75 µs]

Normally, OPTIMOD-PCn will feed transmission channels that do not use preemphasis and you should set this control FLAT. 50µS and 75µS preemphasis are only useful if the OPTIMOD-PCn is protecting a transmission link that uses preemphasis (like certain satellite uplinks) or if you are using a TVxxx preset to drive an analog aural transmitter (i.e., one that produces a preemphasized FM aural carrier in analog television transmission). Use FLAT preemphasis for DAB, DRM, HD Radio, digital television, netcasts, and any other channel that uses a lossy codec. When in doubt, use FLAT preemphasis.

For these applications, you can apply preemphasis to OPTIMOD-PCn’s five-band compressor’s sidechain (to make the five-band compressor “preemphasis-aware,” allowing it to be used as a high-frequency limiter) and before OPTIMOD-PCn’s peak limiter so the limiter will control the peaks of the preemphasized audio.

When the OUTPUT DE-EMPHASIS control is set to FLAT, a deemphasis filter after the look-ahead limiter restores “flat” audio at OPTIMOD-PCn’s output; otherwise, the output is preemphasized. When the output is deemphasized, the transmitter driven by OPTIMOD-PCn must restore the preemphasis before the signal is transmitted.

When in doubt, choose PRE-E and bypass the transmitter’s preemphasis. This prevents potential mismatches between the Optimod’s output deemphasis filter and the transmitter’s input preemphasis filter, which can introduce peak overshoots.

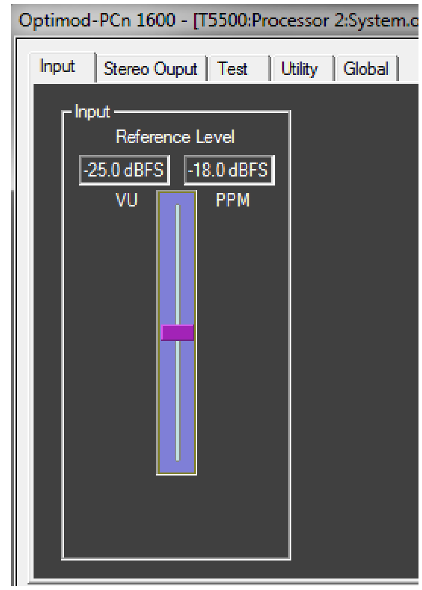
OPTIMOD-PCn’s processing is tuned to be most effective with “flat media” and cannot provide “competitive” loudness for preemphasized radio channels. Because OPTIMOD-PCn 1600 is not sold or licensed as a processor for FM radio transmission, HARD MX OVERSHOOT MODE is locked out when preemphasis is not FLAT. We recommend using one of Orban’s Optimod-FM processors for this application.

OPTIMOD-PCn can process audio for transmission on a preemphasized analog television aural carrier because this application requires lighter processing than FM radio. OPTIMOD-PCn’s TVAxxxx factory presets have been created for this application.

We strongly encourage you to use the MX limiter if you have configured OPTIMOD-PCn to use preemphasis. This will produce far fewer gain-pumping artifacts than will non-MX operation.

If lack of CPU power forces you to use the non-MX limiter, only use it for light-to-moderate protection limiting with a low duty cycle. Otherwise, you may hear pumping on material with a lot of high frequency energy like sibilance and cymbals. The cure is lowering the MB FINAL LIMIT DRIVE control until the problem is no longer audible.

If you find that lowering the MB FINAL LIMIT DRIVE causes too much loudness loss, use the Band 5 compressor as a high frequency limiter to compromise between loudness and limiting artifacts. Set B5 DELTA RELEASE to +6 and B5 STEREO COUPLING to OFF. Adjust B5 THRESHOLD control until you see gain reduction on the Band 5 GR meter with problematic material. Continue to lower the B5 THRESHOLD control until you no longer hear gain pumping. Instead, you will probably hear some high frequency loss. This loss is less subjectively objectionable than gain pumping.



You can use the AGC+LIMITER and TVA factory presets as examples of how to do this. See AGC+LIMIT on page 3-34.

5. Adjust the Input Reference Level control.

This step adjusts the drive to OPTIMOD-PCn's audio processing so that it operates in its preferred range.

A) In the I/O Mixer, click the INPUT tab.

B) Open the GREGG HD MED factory processing preset.

a) On the Optimod-PCn Control application, choose FILE/OPEN PRESET.

The Open Preset dialog box appears.

b) Click GREGG HD MED in the preset list.

The GREGG HD MED preset becomes active. The dialog box remains open until you explicitly close it by clicking DONE.

C) Calibrate using program: Feed normal program material to OPTIMOD-PCn.

a) Play program material from your studio, peaking at the level to which you normally peak program material (typically 0 VU if your console uses VU meters).

b) In the I/O Mixer window, click the GLOBAL tab.

c) Verify that EXTERNAL AGC is set to NO.

Refer to step (2.C) on page 2-16.

d) Click the INPUT tab.

e) Set the INPUT REFERENCE LEVEL control to make the AGC GAIN REDUCTION meter (on the main meter screen in the Optimod-PCn Control application) indicate an average of 10 dB gain reduction when normal levels are applied to the OPTIMOD-PCn's input.

If necessary, scroll the Mixer window horizontally so that the AGC GAIN REDUCTION meter is visible.

If the AGC GAIN REDUCTION meter averages less than 10dB gain reduction (higher on the meter), or if the GATE indicator stays on when program material is present, turn the INPUT REFERENCE LEVEL fader down (so the reference level is lower). f the AGC GAIN REDUCTION meter averages more gain reduction (lower on the meter), turn the INPUT REFERENCE LEVEL fader up.

The VU and PPM calibrations are arranged so that the PPM scale is 7 dB higher than the VU. This approximates the difference in indication between a VU meter and PPM with dynamic program material. See *VU and PPM metering* on page 1-23.

The INPUT REFERENCE LEVEL control is calibrated to comply with industry standard levels. A typical reference level is −18 dBFS (VU) for EBU countries. SMPTE standard reference level is −20 dBFS (VU). If you are using one of those reference level, it is usually OK to simply set the INPUT REFERENCE LEVEL control (VU) to this reference level.

f) When finished, reset AGC to DISABLED, if required.

6. To copy an IO setup from a source Processor to a destination Processor (optional):

a) If you have not already saved the IO setup you wish to apply to the destination Processor, use the 1600PC to connect to the source Processor. Then go to FILE/SAVE SETUP and save the IO setup of the source Processor.

When saving, use any legal operating system filename other than `default.orbs`, which is a reserved name. Setup files have the form `*.orbs`.

A given Processor's setup is stored in the Registry on the computer where that Processor resides.

The current active setup file is always the last file that was recalled by the FILE/OPEN SETUP operation. If you have never recalled a setup file this way, the current active setup is stored in a file named `system.orbs`.

The current active setup file is a transient file. That is, the Optimod-PCn Control application updates it whenever you change the setup manually. The Optimod-PCn Control application also automatically updates the file when the application starts and reads the setup information from the Registry. *You cannot assume that a *.orbs setup file (other than* `default.orbs`*) is static or that it will retain its original information.*

The factory default system setup is stored as `default.orbs`. This is a read-only file. It is the only system setup file that is static and unchanging. For further security, it is automatically regenerated each time the Orban Control Program or Service starts up.

b) Connect the 1600PC to the destination Processor.

c) Go to FILE/OPEN SETUP.

d) Navigate to the folder containing the setup file you wish to retrieve.

This will usually be the file you saved in step (a).

The OPTIMOD-PCn file system labels the Processor folders as "Processor[x]," which is the generic name for a given Processor as shown in the TOOLS > ADMINISTRATION dialog box for that Processor. In that folder will be a preset folder containing the setup you wish to restore.

e) Highlight the setup file and select OPEN.

The Optimod-PCn Control application will automatically make a copy of just-opened setup file in the destination Processor's Presets folder. It will automatically update this file if you make manual changes to the destination Processor's IO setup. The original setup file is not changed.

B) If you wish to edit an existing (or factory) setup, proceed to step 2 below.

C) When you are finished setting up each Processor, close the I/O MIXER window by clicking DONE.

Customizing Processing Presets to Achieve Specified Target Loudness

Radio-Style Presets: Most non-TV presets have local TARGET LOUDNESS values (i.e. TARGET LOUDNESS in the LESS-MORE tab is not set to GLOBAL. We chose these values to represent competitive loudness/distortion trade-offs in traditional, non-R128 applications—their loudness is much higher than −23 LUFS and represents the typical loudness encountered in non-R128 netcasts. (See Figure 2-5 and Figure 2-6 on page 2-19.)

The fact that the presets have local TARGET LOUDNESS is a compromise between the needs of users who process for specified target loudness and those who process more traditionally. You can adjust the loudness of any such preset by changing its TARGET LOUDNESS value to the desired target loudness or to GLOBAL: GLOBAL is convenient for those processing for government-specified target loudness. After you have adjusted TARGET LOUDNESS, LESS-MORE continues to be available and adjusts texture (usually, the amount of gain reduction in the multiband compressor) while changing loudness as little as possible.

Because of their limited graphic resolution, the CBS and BS.1770 loudness meters are calibrated relative to the active TARGET LOUDNESS value: "0" always corresponds to the active TARGET LOUDNESS value. Hence, adjusting TARGET LOUDNESS will not change the loudness meter indications. This makes it easy to customize presets. After you have finished your coloration and texture adjustments, fine-tune the LIMITER DRIVE CONTROL to make the loudness meters indicate "0" on average.

To achieve most dynamic-sounding audio, the CBS and BS.1770 loudness controllers are turned off in the radio-style presets. Therefore, they will produce loudness meter indications that are somewhat less consistent than those produced by the "TV" presets. Nevertheless, they will always provide excellent subjective source-to-source consistency.

TV Presets: The "TV" factory presets do not need customization to achieve correct loudness because out of the box, they provide loudness equal to the global TARGET LOUDNESS value (as set in I/O SETUP > UTILITIES). In almost every application, a factory preset (possibly as customized with LESS-MORE, which preserves correct loudness) will suffice for a given application. However, for the sake of completeness, we provide instructions below describing how to customize them while retaining the correct loudness.

If you are processing for specified target loudness in television or other video applications and wish to customize a factory preset, be aware of two major philosophies, one presented in ATSC Recommendation A/85 and the other in EBU Recommendation R128.

ATSC A/85: If you wish to ensure that dialog levels are consistent from one program segment to the next, use the 1600's CBS loudness meter as a reference for adjusting a user preset's loudness. In this case, we prefer relying on the CBS Loudness Controller to do final loudness control and turning the BS.1770 Safety Limiter off, although you may leave it on if your organization demand strict adherence to BS.1770 meter readings. Consistent loudness of the "anchor element" (usually dialog) is the goal suggested in ATSC Recommendation A/85, which is the basis for CALM Act compliance, and we prefer this goal to strictly relying on the BS.1770 loudness meter.

EBU R128: If you wish to ensure that integrated loudness as measured on a BS.1770-2 meter is consistent from one piece of program material to the next, use OPTIMOD-PCn's BS.1770 loudness meter as a reference. In this case, you may wish to active OPTIMOD-PCn's BS.1770 Safety Limiter, which will constrain BS.1770 integrated loudness with a sliding 10-second integration window from exceeding the setting of the BS.1770 THRESHOLD control. However, note that R128 encourages mixing to achieve a wide dynamic range, where some material may considerably exceed the target loudness when measured on a "short-term" (three-second integration time; ungated) BS.1770 meter. Moreover, R128 requires online loudness control to be defeated if upstream material is known to be pre-processed such that the integrated loudness of each program segment (per BS.1770) is identical to the active Target Loudness value with a ±0.5 LUFS window. To do this, use activate the PASS-THROUGH switch in your active preset (in the LESS-MORE tab) and save the result as a user preset.

See *Appendix A: Using the ITU BS.1770 and CBS Loudness Meters to Measure Loudness* Controller Performance starting on page 3-93 for a comparison of the ATSC A/85 and EBU R128 philosophies of loudness control.

The CBS and BS.1770 loudness meters' calibrations track the 1600's active TARGET LOUDNESS value, which is either (1) the TARGET LOUDNESS value (in the System tab) or (2) the TARGET LOUDNESS value in the active processing preset if this value is not set to GLOBAL.

The controls to which the following procedure refers are described later in Section 3.

A) Make sure that the 1600's active TARGET LOUDNESS value is the same as the dialnorm or Target Loudness metadata you are transmitting to consumers' receivers.

For U.S. program providers, this ensures that the 1600-processed transmission will meet the requirements of the CALM Act.

EBU Tech 3344 has suggestions about setting target loudness when targeting receivers and player devices that cannot receive dialnorm or Target Loudness metadata from the signals they are reproducing.

B) Recall the TV 5B GENERAL PURPOSE preset and set the INPUT REFERENCE LEVEL control in SETUP > INPUT to produce 10 dB of AGC gain reduction with typical input material. Then recall the preset you intend to edit.

C) If the AGC's idle gain (i.e. the gain reduction produced by the AGC when its silence gate is on) is inappropriate, adjust it with the AGC IDLE GAIN control. It is usually adjusted so that the idle gain is equal to the gain reduction that occurs with normal program material. Its normal setting is 0. See *AGC Idle Gain* on page 3-53.

> If you are using a factory processing preset and you have adjusted the input reference level correctly, there is no need to adjust the AGC DRIVE or AGC IDLE GAIN controls.

D) Adjust the MULTIBAND DRIVE to produce the desired amount of multiband gain reduction. We recommend about 5 dB for dialog at normal levels.

E) *If you wish to use the CBS Loudness Controller:* Set the LOUDNESS THRESHOLD control to 0 dB, which matches the CBS Loudness Controller's threshold to the OPTIMOD-PCn's active TARGET LOUDNESS value. Then set the MB LIMIT DRIVE control so that the CBS segment of the LOUDNESS GR meter indicates 3 dB of gain reduction with dialog at normal levels. (The CBS Loudness Controller's gain reduction appears in blue on the LOUDNESS GR meter.)

> You should see the LOUDNESS LEVEL meter peaking around 0 dB when the LOUDNESS GR meter shows that gain reduction is occurring

F) *If you wish to use the BS.1770 Safety Limiter:* Set the BS.1770 THRESHOLD control to your preferred level. 0 LU provides the tightest control as indicated on a BS.1770 meter, but this is likely to cause program material with an usually low peak to RMS ratio to sound too quiet. Most natural sound is produced when the BS.1770 THRESHOLD control is set to +2 LU or higher.

G) *If you do not wish to use the Loudness Controller: S*et the LOUDNESS THRESHOLD and BS.1770 LIMIT THRESHOLD controls to OFF. Then adjust the MB LIMITER DRIVE control to produce an average of 0 LU on the BS.1770 loudness meter with program material at normal levels.

> Presets with the loudness controller off will produce wider source-to-source loudness variation (as indicated on the BS.1770 and CBS loudness meters) than presets with the loudness controller on. However, they will sound more dynamic and natural while still achieving good subjective loudness consistency.
>
> If you chose an appropriate TARGET LOUDNESS value for your transmission, the 1600's limiting meters should rarely indicate any gain reduction. This means that the target loudness is well matched to the headroom in the transmission system.

H) Save your work as a User Preset.

I) Set PASS-THROUGH GAIN (optional).

EBU Recommendation R128 requires online loudness controllers to be defeatable so that they do not change the dynamics of upstream material that is known to meet the R128 requirement: the loudness of each program segment must be within a ±0.5 LU window centered on the active

Target Loudness value. This measurement must be done over the entire program segment using a BS.1770-2 (or higher) "integrated" measurement.

You may operate any processing preset in Pass-through mode; use the PASS-THROUGH SWITCH to enable or defeat pass-though mode and set the PASS-THROUGH GAIN control (in the LESS-MORE tab) as needed.

When set to 0 dB, OPTIMOD-PCn provides unity gain from input to output regardless of the settings of any controls except the PASS-THROUGH GAIN control. The INPUT REFERENCE LEVEL and 100% OUTPUT LEVEL controls do not affect the gain. The normal setting is 0 dB, because this retains the target loudness of the source material.

If you need a pass-through preset you can recall conveniently, set the PASS-THROUGH SWITCH in any factory preset to IN and save the result as a User Preset.

> MX presets have a different input/output delay than non-MX presets. A Pass-through user preset should have the same MX LIMITING control setting as the active preset you use when not in Pass-through mode. We recommend starting with your active preset and then editing it to produce your Pass-through user preset.
>
> To protect the output channel from clipping, the peak limiter remains in-line during Pass-through operation and the 100% OUTPUT LEVEL control operates normally to specify the maximum peak level at the output with respect to 0 dBFS. If you set the PASS-THROUGH GAIN control higher than 0 dB or if the 100% OUTPUT LEVEL control is set lower than 0 dBFS, it is possible that you will observe limiter gain reduction. If you need to use substantial amounts of gain reduction, you will get best results by starting with an MX preset.

Problems and Possible Causes

Always verify that the problem is not the source material being fed to OPTIMOD-PCn, or in other parts of the system.

OPTIMOD-PCn does not start up immediately after Windows boots.

- A delay of 60 seconds is normal and by design. The OptimodService3 start-up mode in Windows is AUTO (DELAYED). This helps prevent Windows from competing with OPTIMOD-PCn for CPU cycles during start-up.

Audio stutters or echoes after Windows is powered up.

- If you are running Windows 8 or higher, you have not defeated Windows Fast Startup and the Service is starting prematurely. See step 1 on page 2-3.

I downloaded a new version of the OPTIMOD-PCn software and it will not run.

- The version of OPTIMOD-PCn software you have purchased never expires, but there is a time limit for free upgrades and technical support. This time limit is burned into your copy protection key or CodeMeter authorization.

To see the time limit, open the SERVICE SETTINGS window from TOOLS > SERVICE SETTINGS and look at the UPGRADE AND SUPPORT RIGHTS TIME LIMIT field. The date shown is the last day that free technical support is available for your license, and you can only run versions of this software released before this date. You may purchase an extension any time before this date by contacting custserv@orban.com. After this date, extensions are no longer available for purchase, and you must purchase a new version of the software if you wish to upgrade.

To support expired licenses, Orban keeps older versions of the software available for download. Choose the newest one that is older than your expiration date.

When you launch the OPTIMOD-PCn control application, meters and controls do not appear.

- You must connect the control application to a Processor using the CONNECT menu. This is true even if the control application and audio processing are running on the same computer.

You cannot connect to a given Processor from 1600PC.

- The Service is not running or is configured incorrectly. Be sure that the output sound device you are using is configured in WINDOWS CONTROL PANEL > SOUND to match OPTIMOD-PCn's output format (stereo or 5.1), sample rate (44.1, 48, 96, or 192 kHz), and bit depth (16-bit or 24-bit). 1600PC will usually issue an error message if there is a mismatch. The Service will not allow audio processing to occur if the input and output devices are configured incorrectly. See step 10 on page 2-8.

- If the Processor is password-protected, you must know it to connect to that Processor. This will be obvious because you will be asked for the password when you try to connect.

- The Service is stopped. The Service will only run if a valid copy protection key is inserted into a USB slot or if the server computer is running CodeMeter authorization software.

- The Service has a build date that is after the expiration date of your software license. See *OPTIMOD-PCn does not start up immediately after Windows boots* above.

- The computer's CPU is insufficiently powerful to support OPTIMOD-PCn operation. This will usually cause audio stuttering, but in extreme cases will cause the Service not to start.

- Necessary runtime files were not installed. To ensure correct installation, you must run the installer application as Administrator.

You cannot connect to the OPTIMOD-PCn Service.

- If the Service has been assigned an Administrator Password (step 9 on page 2-8), you must supply it to gain access to the Service Settings.

- The copy protection key (if used in your version of 1600PCn) is not present. Note that the Service will not start up automatically if you insert the key after you have booted the host computer. If the key was missing, first insert the key and then start the Service from TOOLS > SERVICE SETTINGS in 1600PC.

- If your version of 1600PCn is protected by CodeMeter software authorization, this must be correctly installed and registered. See the document *Instructions to create license request file for Optimod PCn.pdf*, which the 1600PCn installer installs as part of the installation process.

The OPTIMOD-PCn Service crashes when you try to add Processors.

- The computer on which you are running the Service must have a CPU that supports the Intel SSE4.1 vector instruction set. All Intel i-series processors support SSE4.1. See CPU on page 1-4.

- You are attempting to run more processors than your computer's CPU can accommodate. However, this will usually just cause audio stuttering but will not cause the Service to crash. See *Table 1-1: CPU hardware requirements* for one Intel i-series processor hardware core on page 1-5.

You hear audio glitches, pops or clicks.

- Your computer's CPU is insufficiently powerful. See CPU on page 1-4.

- The sample rates of the audio input and output devices are not precisely locked. See *Locking Clocks* on page 1-35.

There is a long-term delay build-up between input and output.

- The sample rates of the audio input and output devices are not precisely locked. See *Locking Clocks* on page 1-35.

You cannot connect to a given OPTIMOD-PCn Processor from a remote computer running 1600PC on your network.

- You must check the ALLOW NETWORK TO ACCESS LOCAL PROCESSORS box in the TOOLS > SERVICE SETTINGS dialog box for the Service on the Service computer that implements that Processor. See *Figure 2-3: Service Settings* on page 2-5.

Other possible causes are:

- The version of 1600PC on the remote computer is the not same as the version on the computer running the Service. If it is not, it will issue an error message. Please install the same version on the remote machine. 1600PC is not copy-protected, so you may freely install it on as many computers as you wish.

- Your computer's firewall is blocking the connection. See page 2-37.

- Password is wrong.

- The target Processor is open on another 1600PC application running on your network.

- The system that houses the target Processor is not on.
- The Orban Windows service routine, `OptimodService3.exe`, is not running in the computer doing the audio processing.
- The Service has malfunctioned. It is usually possible to restart the Service from the TOOLS > SERVICE SETTINGS dialog box in 1600PC running on the Service computer. (You may have to restart it twice.) If all else fails, try restarting `Optimod3Service` from CONTROL PANEL > ADMINISTRATIVE TOOLS > SERVICES.

The Optimix tab does not appear in the User Controls.

- Optimix controls will only appear if the Processor is set for 5.1 output. Setting a given Processor for stereo output automatically locks out Optimix. The stereo synthesizer controls remain, in a tab labeled STEREO SYNTHESIZER.

Hard MX Overshoot mode cannot be activated.

- The HARD MX Overshoot mode is only available when the preemphasis is set to FLAT IN I/O > STEREO OUTPUT.
- HARD MX is a very CPU-intensive process and will cause audio stuttering if the host computer is insufficiently powerful. To reduce CPU load, try defeating Optimix and the CBS Loudness Controller in I/O > UTILITIES. (Note that if the output sound device is stereo, Optimix will be defeated automatically regardless of the indication in I/O > UTILITIES.)

If you have forgotten your Administrator or User Password.

- If you have forgotten an individual Processor's password, you can change it if you know the Administrator Password. See step (12.D) on page 2-11.
- There is no "backdoor" to retrieve a given Service's Administrator password. If you have forgotten the Administrator password, you must uninstall OPTIMOD-PCn using the Orban uninstaller, available from the Windows Programs menu. When you uninstall, you will be asked if you want to retain your settings for use in a new install. Select NO. This will wipe out all of your audio routing settings and passwords, allowing you to create a fresh installation with default settings .

  Before you uninstall, you may use Windows Explorer to make a backup copy of the existing system.orbs for each Processor. The default location is:
  `C:\Users\Public\Documents\Orban\Optimod-PCn 1600\`
  `Processor [x]\presets\`
  Use a different name than `system.orbs` (like `Processor1.orbs`) for the copy. After you reinstall, you can use these files to restore your I/O settings except for I/O routing. Do this by copying them back into the Processor[x] folders after you have connected 1600PC to each of the processors. Then make them active via the FILE > OPEN SETTINGS menu item in 1600PC. You must do this once for each Processor. (See step 6 on page 2-22 for more detail.)

Meters in the 1600PC Control Application freeze momentarily but audio continues to be processed normally.

- This is by design. The software thread controlling the meters is given lower than "normal" priority in Windows to prevent the meters from interrupting important threads that maintain audio continuity. Quickly changing control settings will sometimes temporarily cause the meters to freeze because the control setting messages have a higher priority than the meters.

Poor peak level control

- Thanks to its "true-peak" limiter, which anticipates and controls peak levels following an ideal reconstruction filter in the analog domain, OPTIMOD-PCn audio processing usually controls its output peak levels to an accuracy of 0.2 dB at any output sample rate—in principle, sample rate conversion is similar to reconstruction. However, codecs like HE-AAC have intrinsic peak overshoots and you must allow headroom for these to prevent audible clipping in player devices. 1.5 dB is typically sufficient, as the remaining clipping will have a low duty cycle and is unlikely to be audible. OPTIMOD-PCn's OUTPUT LEVEL control directly controls headroom without affecting loudness. For example, to allow 1.5 dB of headroom for codec overshoots, set this control to −1.5 dB.

- An analog connection can cause analog-domain overshoot if the connection is not phase linear and has a low-frequency cutoff of greater than 0.15Hz (at −3dB).

Audible distortion

- Make sure that the problem can be observed on more than one sound system and at several locations.

- Verify that the source material at OPTIMOD-PCn's audio inputs is clean. Heavy processing can exaggerate even slightly distorted material, pushing it over the edge into unacceptability.

- The subjective adjustments available to the user have enough range to cause audible distortion at their extreme settings. Advancing the MB FINAL LIMIT DRIVE control too far will inevitably cause distortion. (Distortion is very probable if gain reduction in the final limiter frequently exceeds 8 dB.) Setting the LESS-MORE control beyond "9" will cause audible distortion of some program material.

- OPTIMOD-PCn's MX peak limiter has several controls that affect the tradeoff between bass punch and IM distortion between the bass and midrange. Try turning up BASS PRE-LIMITING and BASS LIMITING (move them more towards to right of the screen). Setting BASS PRE-LIMIT MODE to HARD and setting the MX LIMITER THRESHOLD control slightly below "0" can also help.

Audible noise in processed audio

- Excessive compression will always exaggerate noise in the source material. OPTIMOD-PCn reduces this problem with its compressor gate, which freezes the gain of the AGC and compressor systems whenever the input noise drops below a level set by the Gate Threshold control, preventing noise below this level from being further increased.

There are two independent silence gates in each processing channel of the 1600. The first affects the AGC and the second affects the Multiband Compressor. Each has its own threshold control. (See *MB Gate* on page 3-65.)

In sound for picture, the setting of the Gate Threshold control is quite critical if you want the processing to be undetectable to the audience. If this control is set too low, then the 1600 will pump up quiet sounds like ambiance and underscoring to unnaturally high levels. Refer to Section 3 of this manual for a further discussion.

- If you are using OPTIMOD-PCn's with a soundcard having an analog input, the overall noise performance of the system is usually limited by the overload-to-noise ratio of the soundcard's analog-to-digital converter.
- In digital radio applications, if an analog studio-to-transmitter link (STL) is used to pass unprocessed audio to OPTIMOD-PCn, the STL's noise level can severely limit the overall noise performance of the system because compression in OPTIMOD-PCn can exaggerate the STL noise. For example, the overload-to-noise ratio of a typical analog microwave STL may only be 70-75dB. In this case, it is wise to use an Orban Studio Level Controller (like an Orban 6300 or another OPTIMOD-PCn) to perform the AGC function prior to the STL transmitter and to control the STL's peak modulation. This will optimize the signal-to-noise ratio of the entire transmission system. An uncompressed digital STL will perform much better than any analog STL. Section 1 of this manual has a more detailed discussion.

Shrill, harsh sound; excessive sibilance

- Excessively high settings of the HF GAIN control can cause this problem. It can also be caused by excessively high settings of the B5 THRESHOLD (Band 5 Compression Threshold) control. In the latter case, you are first likely to notice the problem as harsh sibilance on voice.

Gain pumping when high frequency energy is present

- This will occur with many OPTIMOD-PCn non-MX factory presets when OPTIMOD-PCn's preemphasis is set to 50 µs or 75µs. [See step 4.C) on page 2-18.] The gain pumping happens because the preemphasis creates a large high frequency boost before the look-ahead limiter, so the look-ahead limiter must produce large amounts of gain reduction to control peak levels.

  To correct this problem, activate the MX limiter (in the DISTORTION tab), and, if necessary, turn down the FINAL LIMITER DRIVE control. Note that activating the MX limiter will cause CPU usage to increase.

  > Compared to the look-ahead limiter that performs peak limiting when the MX limiter is defeated, the MX limiter is much less susceptible to gain pumping.

System will not pass line-up tones at full output level/100% modulation

- This is normal in OPERATE mode. Sine waves have a very low peak-to-average ratio by comparison to program material. The processing thus automatically reduces their peak level to bring their average level close to that of program material, promoting a more consistent and well-balanced sound quality.

  To pass line-up tones transparently, use Bypass Mode (in I/O > TEST) with the BYPASS GAIN set to 0 dB.

  PASS-THROUGH mode will pass tones very close to 100%, but OPTIMOD-PCn's peak limiter remains in the signal path.

Speech/music detector toggles continuously between Speech and Music with a low-frequency test tone

- This can occur when the main highpass filter and speech highpass filters are set differently. If the test frequency is midway between the two filter frequencies, switching between them will cause the tone level to change abruptly, fooling the speech/music detector into thinking that this is a speech waveform. This issue never arises with program material. See *Highpass Filter* on page 3-61.

Dialog is muffled in TV applications

- Set the B3>B4 COUPLING and B4>B5 COUPLING controls to a higher value. This will allow the processing to apply more dynamic high frequency boost. In the surround processing, adjust these controls in the center channel compressor.

- Set the Center (surround) or Speech Mode (stereo) B4 COMPRESSOR THRESHOLD control to a higher value (i.e., closer to 0 dB). This will produce less gain reduction in the presence region.

- Try using the 1600's HF Enhancer (in the EQUALIZER tab) and set the HF ENHANCE control to taste. For most TV programming, moderate settings (like 3 dB) are appropriate because they minimize the possibility of increasing noise.

General dissatisfaction with subjective sound quality

- Make sure that all Windows audio processing is defeated for the Playback Windows sound device you are using. See step 10.A)c) on page 2-9.

- *OPTIMOD-PCn is a complex processor that can be adjusted for many different tastes. For most users, the factory presets, as augmented by the gamut offered by the LESS-MORE control for each preset, are sufficient to find a satisfactory "sound." However, some users will not be satisfied until they have accessed other Modify Processing controls and have adjusted the subjective setup controls in detail to their satisfaction. Such users must fully understand the material in Section 3 of this manual to achieve the best results from this exercise.*

Problems Specific to Dolby Digital Transmissions

Loudness incorrect compared to other Dolby Digital Transmissions

- Be sure that the active TARGET LOUDNESS setting in the 1600 is the same as the dialnorm setting you are transmitting to consumers. Use a device like the Dolby LM100 Loudness Meter to read out the dialnorm you are transmitting to your audience (or simply check the dialnorm setting of your Dolby Digital encoder). See *Customizing Processing Presets to Achieve Specified Target Loudness* on page 2-23.

- Be sure that the 1600's input reference level control is set to produce normal amounts of gain reduction. See step 5 on page 2-21.

- If the loudness controller is on, be sure that the LOUDNESS THRESHOLD control is set to 0 dB.

  This matches the loudness controller's threshold to the 1600's active TARGET LOUDNESS value.

- *If you are using the loudness controller:* Once Target Loudness is correct, be sure that your active preset is causing the 1600's Loudness Gain reduction meter to indicate approximately 3 dB of gain reduction on normal dialog. Adjust its MB LIMIT DRIVE control if it does not.

  There are two LOUDNESS LEVEL meters. "dB" indicates according to the CBS (Jones & Torick) algorithm. The Lk (or LU) meter indicates BS.1770 short-term loudness as a floating yellow segment and BS1770-3 Integrated Loudness as a bar. The Integrated measurement uses a 10-second rolling window.

  When the loudness controller is operating normally, the 1600's LOUDNESS LEVEL (dB) meter should be peaking around 0 dB on dialog. If it is not and you are using a custom preset, you might get better loudness control by slightly tweaking the LOUDNESS THRESHOLD control to make the LOUDNESS LEVEL (dB) meter peak around 0, bearing in mind that the correct setting is 0 dB for typical sound-for-picture processing. A maximum variation of ±1 dB will typically suffice.

  If the 1600's active TARGET LOUDNESS setting is not the same as the dialnorm setting you are transmitting to your audience, the 1600's Loudness Level meters will be calibrated incorrectly, so even if the meters are indicating 0 dB, loudness at the consumer's receiver will not be correct.

  Note that because the 1600's Loudness Level (dB) meter shares the loudness controller's filterbank, the meter does not show the effect of the LOUDNESS ATTACK control, which shapes the loudness controller's gain reduction signal outside the loudness controller's feedback loop. It also does not show loudness reduction caused by OPTIMOD-PCn's peak limiters Therefore, if you are using values of LOUDNESS ATTACK below about 50% and/or large amounts of peak limiting, the loudness of transient events may be significantly higher than the Loudness Level (dB) meter indicates.

The BS.1770 meters [LOUDNESS LEVEL (Lk or LU)] are located after all processing, so they accurately indicate the BS.1770 short-term and Integrated loudness of OPTIMOD-PCn's output.

- *If you have defeated the loudness controllers:* If you have made a custom preset with the loudness controllers defeated, use the FINAL LIMIT DRIVE control to calibrate the preset's loudness such that the BS.1770 loudness meter indicates 0 LU during normal operation with typical program material. Loudness will not be as consistent as it is when the loudness controller is active.

A BS.1770 loudness meter indicates higher loudness than expected with some program material

- Set the BS.1770 SAFETY LIMIT THRESHOLD control closer to 0 LU. This control is in the active processing preset. Using the BS.1770 safety limiter will prevent the meter from reading above the preset threshold but may cause subtle loudness pumping because of limitations in the BS.1770 algorithm. See *BS.1770 Safety Limiter* on page 3-26 for a more detailed discussion.

  Your Optimod has a built-in BS.1770-2 (or higher) long-term (gated) meter with a 10-secondintegration time. This should agree with an external BS.1770-2 meter that is set up with a 10-second gated integration time, a "Relative" scale (i.e. Lk or LU), and a reference level equal to your Optimod's active TARGET LOUDNESS value.

- We do not recommend using the loudness controllers with music-oriented formats, as they will unnecessarily constrain dynamics. In particular, the BS.1770 Safety Limiter is likely to cause music produced with high amounts of dynamics compression to be unnaturally quiet with respect to speech. If the loudness controllers are turned off (as they are in most radio-style presets), the setting of the TARGET LOUDNESS control will correspond only approximately to the measured loudness. To fine-tune the loudness for your program material, tweak the MB FINAL LIMIT DRIVE control. Use OPTIMOD-PCn's loudness meters to guide you.

CBS Loudness Controller reduces transient punch of programming

- Reduce the amount of fast gain reduction in the loudness controller by setting the LOUDNESS ATTACK control closer to 0%. This will allow more short loudness peaks to pass through without attenuation by the loudness controller. We believe that the range from 50% to 70% offers the most useful tradeoffs between reducing punch and allowing irritating short-term loudness bursts to pass through uncontrolled.

Transient loudness events (like esses in speech) sound obtrusively loud

- Set the SPEECH B5 THRESHOLD control more negative.

- Set the TRANSIENT ENHANCE control closer to 0 ms.

Commercials too loud in sound for picture applications

- Make sure that the Loudness Controller is activated on the preset that you are using—the LOUDNESS THRESHOLD control must not be set OFF (see page 3-72) and is normally set to 0 dB.

- If the Loudness Controller is active but based on its gain reduction meter, you do not believe it is working hard enough, set its threshold lower using the LOUDNESS THRESHOLD control or increase its gain reduction by turning up the MB FINAL LIMIT DRIVE CONTROL, which is located before the loudness controller.

  Note that the Loudness Controller controls subjective loudness to an absolute threshold and does not understand the context of the program. Therefore, if a commercial follows a piece of very quiet program material, the commercial may still seem loud even though the Loudness Controller is working properly.

  See Loudness Control on page 3-24.

Technical Support

If you require technical support, contact Modulation Index customer service (but please try to find the answer in this manual first). Free technical support is available during your support period. In 1600PC, you can see when your support period expires in the *Upgrade and Support Rights Time Limit* field in the TOOLS > SERVICE SETUP window. (See *About the Upgrade and Support Rights Time Limit Field* on page 2-6.)

See https://www.indexcom.com/about/ for contact information. Be prepared to describe the problem accurately. Know the serial number of your 1600 and the software, and service versions are you running. The Windows operating system version and hardware platform will also be required.

> The software version number is available in 1600PC via Help > About Optimod-PCn 1600… If you used the automatic installer, the software version number in the 1600PC installed in the computer running the Service will be the same as the version number of the Service.

Before you return a product to the factory for service, please refer to this manual. Make sure you have correctly followed installation steps, operation procedures, and any appropriate troubleshooting suggestions. If you are still unable to solve a problem, contact our Customer Service department. Often, a problem is relatively simple and can be fixed quickly after telephone or email consultation.

If you must return a product for factory service, please notify Customer Service by telephone or email *before* you ship the product; this helps us to be prepared to service your unit upon arrival. In addition, when you return a product to the factory for service, we strongly recommend you include a letter describing the problem.

Please refer to the terms of your Limited One-Year Standard Warranty, which extends to the first end user. After expiration of the warranty, a reasonable charge will be made for parts, labor, and packing if you choose to use the factory service facility. Returned units will be returned C.O.D. if the unit is not under warranty. Orban will pay return shipping if the unit is still under warranty. In all cases, the customer pays transportation charges to the factory (which are usually quite nominal).

Adding a Custom Logo

A user defined bitmap can be displayed at the right of the meter dialog in the PC Control application. There is no default bitmap assignment available. You may create one bitmap that works with all Processors hosted by a given Service, or you can use individual bitmaps for each Processor.

The Control Application will not automatically scale the bitmap size. It must be created with these parameters:

Size: 180 pixels wide x 90 pixels high
Format: Windows 24-bit
Filename: logo.bmp

To use this feature, create or copy a 180 x 90 pixel standard Windows format .bmp file and name it `logo.bmp`. If you wish this to appear on all Processors, place this file in the `presets` directory. This will usually be the default application install directory:

Windows 32-bit
`c:\Program Files\Orban\OPTIMOD-PCn 1600\Processor[x]\presets\`

Windows 64-bit
`c:\Program Files (x86)\Orban\OPTIMOD-PCn 1600\Processor[x]\presets\`

If you wish to have a different logo for each Processor, do not put logo.bmp in Program files. (If you do, it can overwrite the logo.bmp files for the individual Processors.) Instead, put logo.bmp here (where Processor x is the name of the Processor):

`C:\Users\Public\Documents\Orban\Optimod-PCn 1600\Processor x\presets`

Using Windows Firewall with OPTIMOD-PCn

Depending upon your network security requirements, it may be necessary to configure a software firewall application like Windows Firewall, included with Windows.

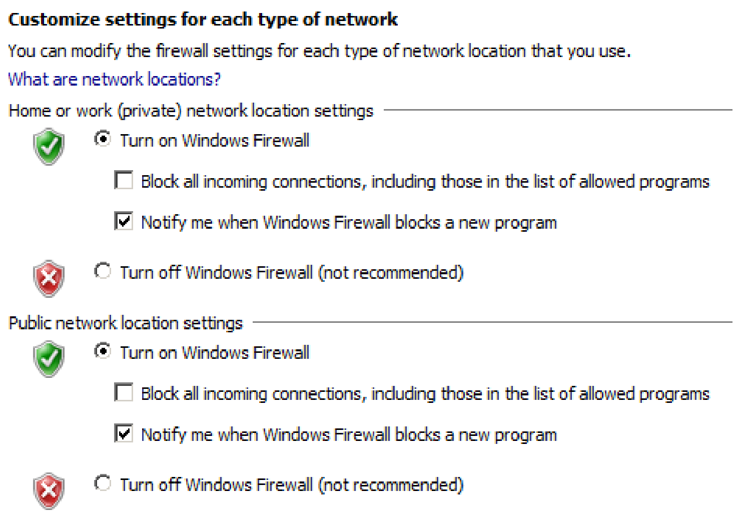
1. **Enable Windows Firewall:**

   A) Navigate to CONTROL PANEL > WINDOWS FIREWALL

   B) Click `Turn Windows Firewall On or Off`.

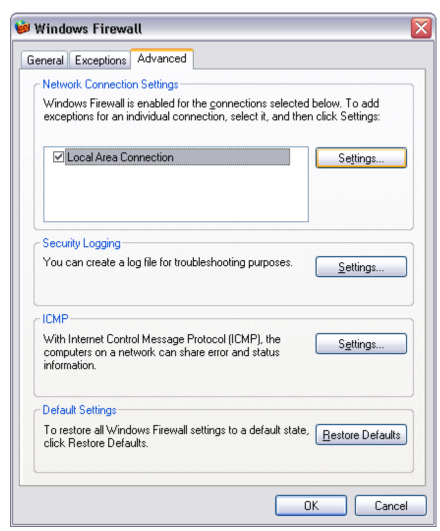
   C) Enable the On (recommended) button.

   D) Select the Advanced tab.

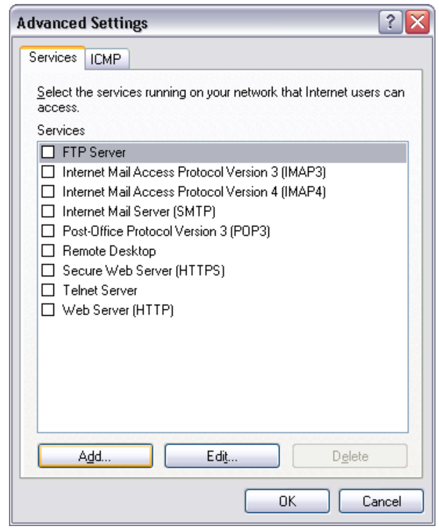


2. **Choose the network device that will be used for OPTIMOD-PCn access and will be protected with Windows Firewall.**

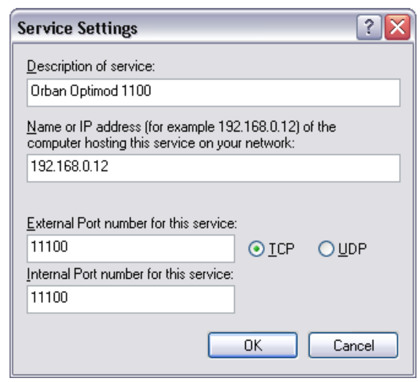
   A) Click Settings…



B) Click Add…



C) Enter the Description of the service, the name or IP address, the External Port number, and the Internal Port number.



D) Select TCP.

The IP address will be that of the computer. To avoid any conflict, we recommend choosing a Port higher than 1024 that is not otherwise assigned according to IANA:

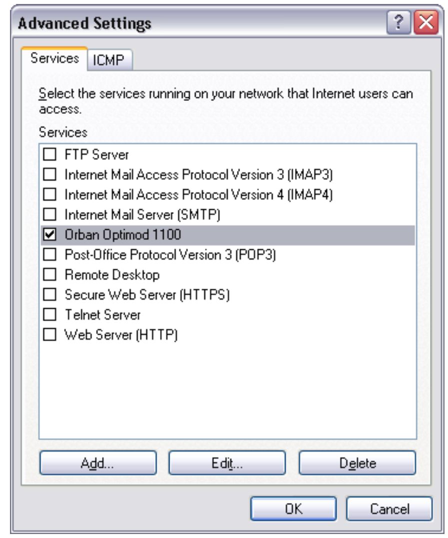
http://www.iana.org/assignments/port-numbers

- Port 12100 is the default for the OPTIMOD-PCn Control Application.
- Port 12101 is the default for the OPTIMOD-PCn Terminal Application.

All processors are controlled by the same Service and share the sma port number.

E) Select OK.

A dialog box appears, indicating that the Orban Optimod 1600 Service is accessible through Windows Firewall.

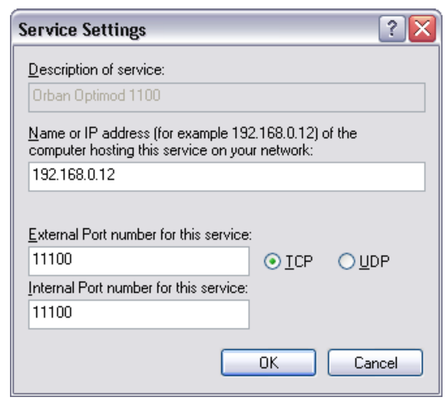


F) If any changes are required (like IP address or port):

a) Click Edit.

b) Make the changes.

c) Click OK.



NAT (Network Address Translation) Firewalls

If the computer containing OPTIMOD-PCn is on a network behind a NAT firewall, you must configure the firewall to allow outside access to the OPTIMOD-PCn computer from another computer outside the firewall. This is usually done via router port forwarding: The OPTIMOD-PCn port(s) are forwarded to the OPTIMOD-PCn computer by configuring the NAT firewall accordingly. The OPTIMOD-PCn computer is then addressed with the network IP address (WAN) instead of the actual IP address of the OPTIMOD-PCn computer.

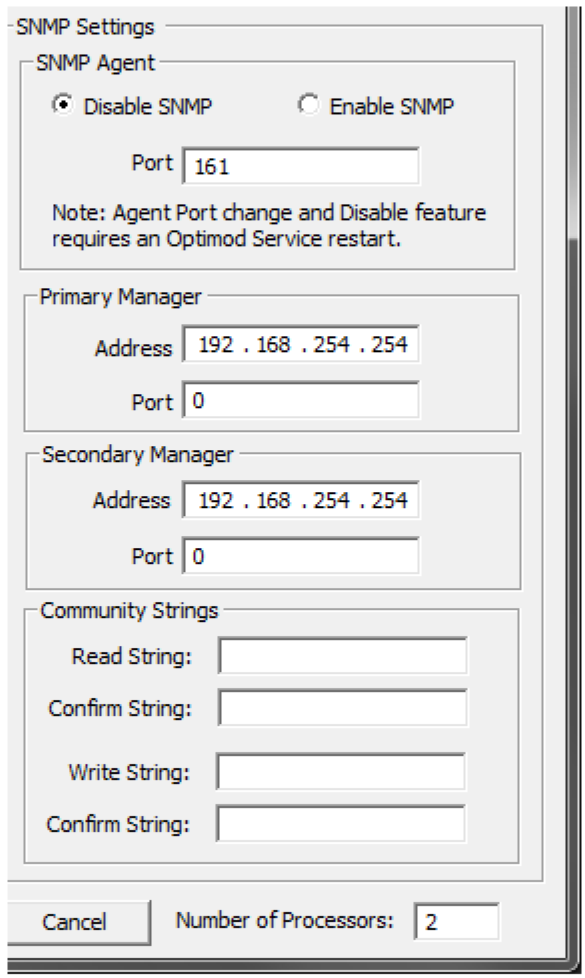
For a single IP addressable network, each port can only be used once unless your router supports port forwarding. Therefore, if you have more than one computer with OPTIMOD-PCn requiring outside access, each computer must have a unique OPTIMOD-PCn port. If your router supports port forwarding, you can assign unique incoming ports, map them to the different IP addresses and then use the same ports on each computer.

The OPTIMOD-PCn Control application uses TCP only.

Refer to your router/firewall documentation for the exact configuration procedure, paying close attention to the network security risks inherent in configuration changes.

SNMP Support

[SNMP is not supported in v0.9x software.]



The SNMP (Simple Network Management Protocol) features allow you to monitor your Optimod's status and to send Alarm notifications to your network via the Ethernet connection of your Optimod's host computer. It is beyond the scope of this manual to provide a general explanation of how SNMP works. The text below provides sufficient information to use your Optimod in your specific SNMP setup if you are already familiar with the general principles of setting up SNMP.

SNMP Software Installation

SNMP support is installed automatically on your computer when you install OPTIMOD-PCn software.

SNMP Network Setup

To set up SNMP, you must run OPTIMOD-PCn control software on the same computer that houses the OPTIMOD-PCn card(s) being monitored by SNMP. SNMP runs as part of the same Windows Service that manages communication between OPTIMOD-PCn, your computer, and your network.

From the TOOLS menu, select SERVICE SETTINGS to access the SNMP configuration controls.

- **SNMP (Enable/Disable): enables or disables the SNMP feature.**
- **Primary Manager Address: sets the address of the Primary SNMP Manager.**
- **Primary Manager Port: sets the port of the Primary SNMP Manager.**

  When SNMP is enabled, you can disable the primary or secondary manager by setting its port to 0.

- **Secondary Manger Address: sets the address of a Secondary SNMP Manager.**

- **Secondary Manger Port: sets the address of a Secondary SNMP Port.**

SNMP mib File Location

The default 1600 install location is:

`\Program Files\Orban\OPTIMOD-PCn 1600\orban1600.mib`
or
`C\Program Files(x86)\Orban\OPTIMOD-PCn 1600\orban1600.mib`

SNMP Default Settings

- SNMP Agent: Disabled
- Primary Manager(Alarm) Address: 192.168.254.254
- Primary Manager (Alarm) Port: 0
- Secondary Manger (Alarm) Address: 192.168.254.254
- Secondary Manger (Alarm) Port: 0

SNMP Features

Get/Query:

- Processor name
- Software version
- Primary and Secondary Manager IP
- Primary and Secondary Manager Port
- InputSilence

Traps/Alert:

- InputSilence

SNMP Community String:

The “SNMP Community string” is like a user id or password that allows access to a router's or other device's statistics. Set it in SERVICE SETTINGS>SNMP SETTINGS to implement SNMP security. PRTG sends the community string along with all SNMP requests. If the community string is correct, the device responds with the requested information. If the community string is incorrect, the device simply discards the request and does not respond.

- Read String allows users to set a password for SNMP to retrieve information from the Optimod. Default is PUBLIC.
- Write String allows users to set a password for SNMP to write (set) information from the Optimod. Default is PRIVATE.

Section 3

Operation

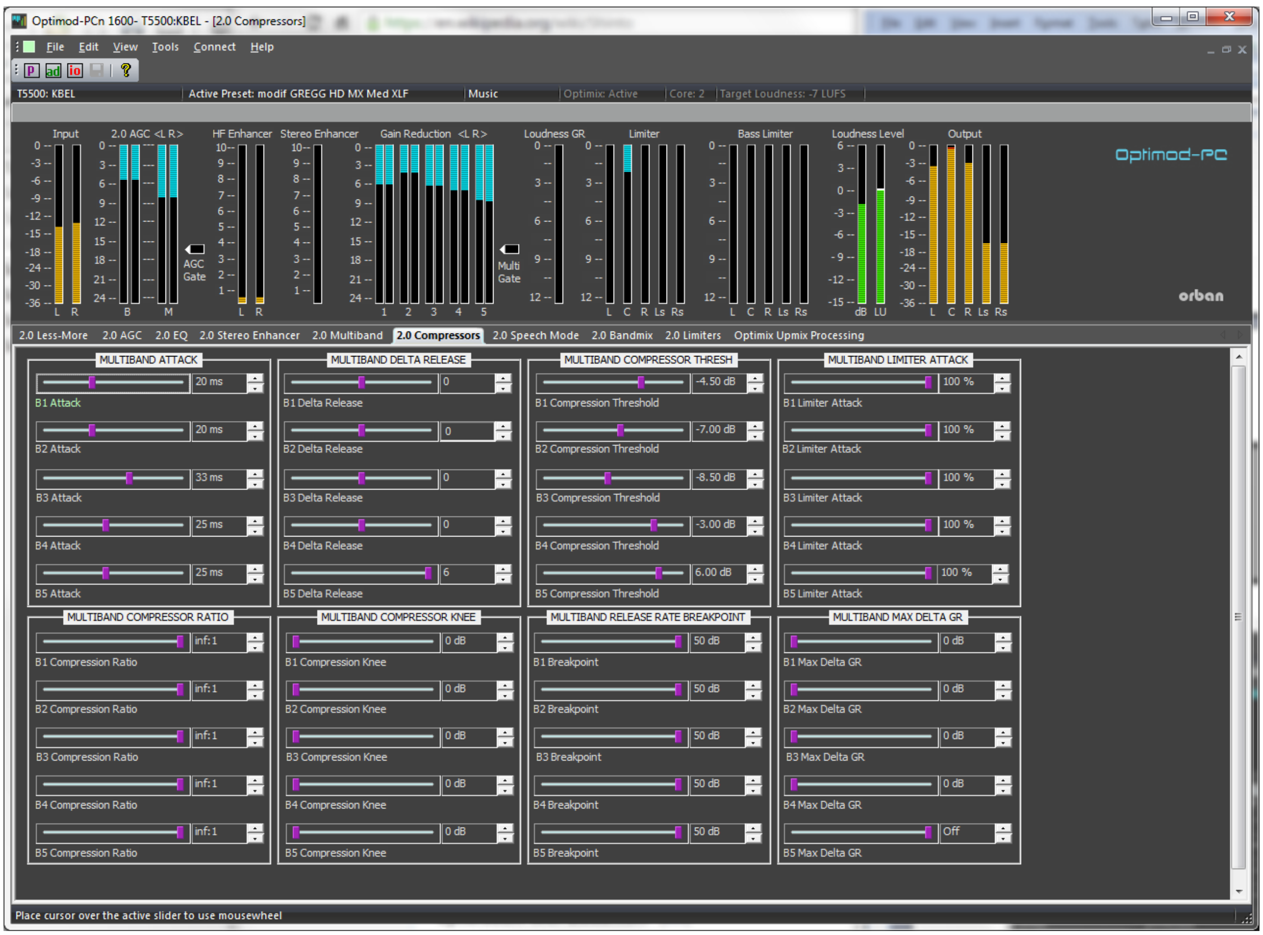


Figure 3-1: The OPTIMOD-PCn Control Application

The OPTIMOD-PCn Control Application

- **AGC** meter shows the gain reductions of the "Master" (above-200 Hz) and "Bass" (below-200 Hz) bands in the slow AGC processing that precedes the multiband compressor. Full-scale is 24dB gain reduction.

- **Gate indicators** show gate activity. They light when the input audio falls below the threshold set by the gate threshold controls. (There are two gating circuits—

one for the AGC and one for the multiband compressor/limiter—each with its own gate threshold control.) When gating occurs, the AGC and compressor's recovery times slow drastically to prevent noise rush-up during low-level passages.

- **Stereo Enhancer meter** shows the amount of extra gain, in dB, being applied to the stereo difference signal (L−R) to increase the apparent width of the stereo soundstage.

- **HF Enhancer meter** shows the amount of dynamic 4 kHz first-order high frequency shelving equalization, in dB, being applied to the signal.

- **Multiband gain reduction meters** show the gain reduction in the multiband compressor. Full-scale is 25 dB gain reduction. In a stereo instance, each meter is split vertically to show the gain reduction in the left and right channels. In a surround instance, each meter is split vertically to show the gain reduction in the main (all channels but center) and center channels. See *Multiband Compressor/Limiter* starting on page 3-19 for an explanation of how the main and center compressors are coupled in surround instances.

- **Input Meters** show the peak input level applied to the audio processing, in units of dBFS. Higher levels call for higher settings of the INPUT REFERENCE LEVEL control. If the meter indicates "0," this indicates that the input peak level has reached 0 dBFS. This is common if the program material applied to OPTIMOD-PCn was aggressively mastered and there is unity gain between the playout system and the input of OPTIMOD-PCn.

- **Output Meters** show the peak level of the processed samples at OPTIMOD-PCn's output, in dBFS. They will clearly show the effect of the 100% OUTPUT LEVEL CONTROL. (For example, setting this control to −2.0 dBFS will cause the meters to peak at −2.

  If the output of a stereo Processor is configured for 5.1-channel, five output meters will appear to show the effect of the Optimix stereo→surround upmixer. Note that there is no LFE channel shown because Optimix places energy below 80 Hz in the Lf and Rf channels instead of in an LFE channel. This makes it maximally compatible with home theater receivers' bass management functionality—if Lf and Rf speakers are "small," the receiver will route bass to the LFE channel if present.

- **Limiter** meters show the amount of peak limiting (dB) in the various audio channels, which we chose not to couple because the fast release time of this circuit would otherwise cause elements in one channel to modulate the opposite channel objectionably. Full-scale is 12 dB gain reduction.

  If the output of a stereo Processor is configured for 5.1-channel, five limiter meters will appear.

- **Bass Limiter** meters show the amount of peak limiting applied to the bass energy in the various audio channels. Full-scale is 12 dB gain reduction.

  If the output of a stereo Processor is configured for 5.1-channel, five limiter meters will appear.

- **Loudness** meters shows the subjective loudness of the output, measured using the 1981 Jones & Torick CBS Technology Center algorithm and by the ITU. BS.1770 algorithm. (See page 1-26 for a discussion of the CBS meter and page 1-17 for BS.1770.)

  > There is only one meter for each algorithm regardless of the number of channels and loudspeakers in the listening room because a given listener has only one perception of loudness.
  >
  > The meter is calibrated with reference to the Target Loudness / dialnorm value that you specify in the 1600's active Setup for that processing chain. (See step 4.C) on page 2-18.) If you adjust the processing so that the Loudness Level meter peaks at 0 dB on dialog and you set up your Dolby Digital encoder so that that you are transmitting this same dialnorm metadata value to consumer receivers, your transmission will have the correct loudness compared to other correctly set up transmission channels.
  >
  > If the CBS loudness controller is defeated in I/O MIXER > UTILITY, the CBS Loudness Meter (dB) will show no activity.

- **Loudness Gain Reduction** shows the amount of gain reduction that the Loudness Controller and BS.1770 safety limiter are producing. The CBS controller gain reduction appears in blue, while the BS.1770 safety limiter gain reduction appears in cyan. The two indications are stacked, so the edge of the meter shows the total gain reduction produced by both algorithms. Full-scale is 12 dB gain reduction. These meters will only show gain reduction if the LOUDNESS THRESHOLD and BS.1770 SAFETY LIMIT THRESHOLD are not off (MULTIBAND tab in the active processing preset), and both loudness controllers are activated in I/O MIXER > UTILITY.

- **Downmix meters** (in surround processors) indicate the amount of loudness control, peak limiting, and high frequency limiting produced by the downmix processing. The high frequency limiter is only active when the downmix operates preemphasized, as set in SETUP > OUTPUT. The downmix loudness controller is only active in surround instances of the processing.

- **Control Pane** shows editing controls that allow you to customize the factory presets. There are two levels of control: Basic Control and Advanced Control. The control pane is organized in tabs. Clicking a given tab will bring up the controls pertaining to the tab's title.

  Sliders can be grouped according to the following rules. When sliders are grouped, adjusting one slider will cause all other sliders in that group to move by the same number of increments.

- At least one slider in a given window is always active and will be the base for any grouping.
- SHIFT CLICK will toggle a slider's status (ADD or REMOVE) with the current group of one or more sliders.
- CONTROL CLICK will add a slider to the current group.
- CLICK DRAG will un-group all current sliders and make a new group consisting of the sliders within the hatched selection box.
- CONTROL CLICK DRAG will add the sliders within the hatched-selection box to the current group
- CLICKing on a part of the window outside of any slider will ungroup the sliders.
- CLICKing a slider outside the group will ungroup all current sliders and select the clicked one.

- **I/O Window** allows you to set input and output levels and system setup parameters.

- **File Menu** allows you to open factory and user presets and to save user presets that you have created by editing factory presets or older user presets. When you save a preset, it is saved on the computer housing the OPTIMOD-PCn card on which you created the preset and to the computer running the Control Application. To share presets between cards, use a file manager or Windows Explorer to copy preset files from one folder to another.

  You can also save and restore the state of the OPTIMOD-PCn I/O controls by using the SAVE SETUP and OPEN SETUP menu items.

- **Edit Menu** can open up the OPTIMOD-PCn I/O Mixer screens. It can reveal Advanced Control, which is organized in tabs and allows you to edit processing presets to get the sound you want.

- **View Menu** allows you to display or hide the Toolbar, which contains icon-based shortcuts for common tasks. It also allows you to hide or display the status bar and the Connection List, and lets you activate help for individual controls using Windows Tooltips.

- **Tools Menu** allows you to access the OPTIMOD-PCn I/O window, the PROCESSOR ADMINISTRATION screen, and the SERVICE SETTINGS Screen.

- **Connect Menu** allows you to connect to an OPTIMOD-PCn Processor to perform the various tasks implemented by the Control application and the OPTIMOD-PCn Mixer application. It also allows you to add, edit and remove profiles and to disconnect from a card.

- **Help Menu** provides access to the Help and About functions, including the Operating Manual.
- **Info Bar** shows:
  - the OPTIMOD-PCn Processor to which you are currently connected
  - the preset that the Processor is running
  - the state of the automatic Music/Speech detector
  - the state of the Optimix upmixer (ACTIVE, INACTIVE, or DEFEATED)
  - the CPU core in which the Processor is running
  - the active TARGET LOUDNESS value.
- **Toolbar** contains icons that implement common functions, like recalling and saving presets, opening the I/O mixer screen, opening the advanced modify screens, saving presets, and Help.

Introduction to Processing

Some Audio Processing Concepts

Reducing the peak-to-average ratio of the audio increases loudness. If peaks are reduced, the average level can be increased within the permitted modulation limits. The effectiveness with which this can be accomplished without introducing objectionable side effects (such as pumping or intermodulation distortion) is the single best measure of audio processing effectiveness.

Compression reduces the difference in level between the quiet and loud sounds to make more efficient use of permitted peak level limits, resulting in a subjective increase in the loudness of quiet sounds. It cannot make loud sounds seem louder. Compression reduces dynamic range relatively slowly in a manner similar to riding the gain. Limiting and clipping, on the other hand, reduce the short-term peak-to-average ratio of the audio.

Limiting increases audio density. Increasing density can make loud sounds seem louder, but can also result in an unattractive busier, flatter, or denser sound. It is important to be aware of the many negative subjective side effects of excessive density when setting controls that affect the density of the processed sound.

Clipping sharp peaks does not produce any audible side effects when done moderately. Excessive clipping will be perceived as audible distortion.

Look-ahead limiting is limiting that prevents overshoots by examining a few milliseconds of the unprocessed sound before it is limited. This way the limiter can anticipate peaks that are coming up.

> In Dolby Digital transmission channels, appropriate setting of the TARGET LOUDNESS metadata parameter will allow enough headroom to keep peak levels below the threshold of the 1600's peak limiters. The best sounding limiting is no limiting at all.

Loudness control prevents the subjective loudness from exceeding a preset threshold.

Distortion in Processing

In a competently designed processor, distortion occurs only when the processor is controlling peaks to prevent the audio from exceeding the peak modulation limits of the transmission channel. The less peak control that occurs, the less likely that the listener will hear distortion. However, to reduce the amount of peak control, you must decrease the drive level to the peak limiter, which causes the average level (and thus, the loudness) to decrease proportionally.

Loudness and Distortion

In processing, there is a direct trade-off between loudness and distortion. You can improve one only at the expense of the other. Thanks to our psychoacoustically optimized designs, this is less true of Orban processors than of any others. Nevertheless, all intelligent processor designers must acknowledge and work within the laws of physics as they apply to this trade-off.

> In AM and FM processing, we have long said that there is a direct trade-off between loudness, *brightness*, and distortion, However, because DAB and netcasting systems don't use preemphasis, there is no problem getting the audio to sound bright and the trade-off is only between loudness and distortion.

Perhaps the most difficult part of adjusting a processor is determining the best trade-off for a given situation. We feel that it is usually wiser to give up ultimate loudness to achieve low distortion. A listener can compensate for loudness by simply adjusting the volume control. However, a listener cannot make an excessively compressed or peak-limited signal sound clean again.

If processing for high quality is done carefully, the sound will also be excellent on small playback systems. Although such a signal might fall slightly short of ultimate loudness, it will tend to compensate with an openness, depth, and punch (even on small speakers) that cannot be obtained when the signal is excessively squashed.

If women form a significant portion of the station's audience, bear in mind that women are more sensitive to distortion and listening fatigue than men. In any format requiring long-term listening to achieve market share, great care should be taken not to alienate women by excessive stridency, harshness, or distortion.

Speech/Music Detector

The Speech/Music Detector allows OPTIMOD-PCn to change its processing parameters depending on whether the input program material is speech or other material (usually music).

The algorithm is straightforward: Speech is detected if (1) the input is mono, and (2) there are syllabic pauses at least once every 1.5 seconds. Speech with a stereo music background will usually be detected as "music," or the detector may switch back and forth randomly if the stereo content is very close to the stereo / mono detector's threshold. Mono music with a "speech-like" envelope may be incorrectly detected as "speech." Music incorrectly detected as "speech" can exhibit a slight loss of loudness and punch, but misdetection will never cause objectionable distortion on music.

Speech that is not located in the center of the stereo sound field will always be detected as "music" because the detector always identifies stereo material as "music."

Processing for Audio Codecs

Professional netcasters committed to providing their audiences with the best audio possible at a given bitrate should consider the use of a high-performance codec like Modulation Index's StreamS, which uses standards-based MPEG-4 AAC/HE-AAC/He-AACv2/xHE-AAC. xHE-AAC is the most efficient codec available at the time of this writing. It can provide good entertainment-quality audio at bit rates as low as 24kbps stereo or 16kbps mono, allowing coverage of audiences listening on dial-up connections or via wireless devices. The lower bitrates also penetrate crowded networks with fewer audio interruptions.

OPTIMOD-PCn's ability to maintain source-to-source spectral consistency is also an important advantage. Once you have set up the processing to minimize codec artifacts caused by a given piece of program material, OPTIMOD-PCn's will automatically minimize codec artifacts with any program material.

All codecs add peak overshoot to the audio. This because they remove energy that is psychoacoustically masked by the input audio. It is not unusual for low bit rate codecs to overshoot by 2 dB. To minimize the possibility of clipping at the decoder, it is wise to allow 2 dB of peak headroom at the encoder. In other words, set OPTIMOD-PCn's output level control to −2 dBFS when driving an audio encoder. While there may be a few low-energy overshoots above this, clipping them will not cause audible artifacts.

Depending on the type of program material you are processing, at lowest bitrates (32 kbps and lower with HE-AAC) it is may be advisable to use presets with OPTIMOD-PCn's MX limiter defeated because the extreme peak density produced by the MX limiter may unduly stress the codec. However, defeating the MX limiter will also reduce punch and transient definition, so you must decide by careful listening to the output of the decoded signal after the codec's encode/decode cycle.

Some decoders contain a peak limiter that prevents clipping if enabled. However, the subjective quality of any such peak limiter is unpredictable to the netcaster, so it is better to avoid activating them at all.

OPTIMOD-PCn in Radio-Oriented Applications: From Bach to Rock

OPTIMOD-PCn can be adjusted so that the output sounds:

- as close as possible to the input at all times (using the Two-Band Protection Limiter preset)
- open but more uniform in frequency balance (and often more dramatic) than the input (using the Two-Band structure or running the Five-band structure with slow release time)
- dense, quite squashed, and very loud (using the Five-band structure with faster release times)

The dense, loud setup will make the audio seem to jump out of car and table radios, but may be fatiguing and invite tune-outs on higher quality home receivers. The loudness/distortion trade-off explained above applies to any of these setups.

In professional broadcasting environments, you will achieve best results if Engineering, Programming, and Management go out of their way to communicate and cooperate with each other. It is important that Engineering understand the sound that Programming desires, and that Management fully understands the trade-offs involved in optimizing one parameter (such as loudness) at the expense of others (such as distortion or excessive density).

Never lose sight of the fact that, while the listener can easily control loudness, he or she cannot make a distorted signal clean again. If such excessive processing is permitted to audibly degrade the sound of the original program material, the signal is irrevocably contaminated and the original quality can never be recovered.

Sound for Picture Applications: Controlling Dynamic Range

The most crucial commandment in sound for picture is this: *dialog must always be intelligible.* Sound for picture is usually heard under less-than-ideal conditions and its dynamic range must be controlled accordingly. Apartment-dwellers must set their volume controls to avoid disturbing neighbors or even other members of the family. At the quiet side, intelligibility of dialog is often impacted by environmental noise such as children playing or a dishwasher going in the kitchen. When one considers that the hearing acuity of a significant portion of the audience is somewhat impaired compared to that of a healthy 20-year-old, one concludes that *the dynamic range of dialog must not exceed 15dB* if it is to be intelligible to 99% of viewers under common domestic viewing conditions. Feature-film dynamic range is inappropriate for home viewing (except in dedicated home theaters) and the dynamic range of a significant portion of the audio from video source material must be compressed to best serve the audience. The challenge (which OPTIMOD-PCn effectively meets) is to compress dynamic range unobtrusively.

OPTIMOD-PCn can be adjusted so that the output sounds as close as possible to the input at all times (using the PEAK LIMITER preset), *or* so that it sounds open but more uniform in frequency balance than the input (using considerable interband coupling and a slow release time in the five-band compressor), or so that it sounds dense, quite squashed, and very loud (using faster release times and less band coupling).

In television audio, inconsistent loudness between channels or program elements is annoying, so the dense, loud setup is never appropriate. The 1600 offers two-band and five-band presets (whose names begin with "TV") that exploit the AGC's and multiband compressor's compression ratio controls to subtly control dynamic range in sound for picture applications. These presets effectively and unobtrusively maintain dialog intelligibility while retaining a sense of dynamic range, allowing low-level elements to be heard easily. Meanwhile, the CBS Loudness Controller prevents subjective loudness from exceeding a preset ceiling.

The preset tuning controls on the 1600 give you the flexibility to adapt the processing to individual program segments. In most cases, your goal should be to choose the type of processing that best optimizes dynamic range while controlling the loudness of the loudest sounds so that they are not irritating and are consistent with the loudness of other stations or sources. When the 1600 is otherwise set up correctly (so that it is cognizant of the dialnorm metadata you are transmitting to viewers), its TVxxxx presets achieve this goal most precisely by exploiting the loudness controller. The "radio-style" presets are crafted to match the target loudness as well as can be achieved without use of the loudness controller.

> If you want more consistent loudness from a "radio-style" preset, set its LOUDNESS CONTROLLER THRESH control to −10 dB, turn up its MB LIMITER DRIVE control until the loudness controller gain reduction meter indicates about 3 dB of gain reduction with typical program material, and save the result as a user preset. However, this will often reduce punch with musical programming.

Protection Limiting

OPTIMOD-PCn has an advanced peak limiter that is more than competitive with the best mastering peak limiters, particularly in MX mode. The PEAK LIMITER MX preset allows you to use OPTIMOD-PCn as a mastering-style limiter that is capable of a surprisingly favorable trade-off between loudness and undesired artifacts. It is typically possible to make −6 or −7 LUFS masters that sound clean, punchy, and open.

Studio AGC

You can use OPTIMOD-PCn as a studio AGC and STL protection peak limiter. In a typical application, OPTIMOD-PCn substitutes for the AGC in an Optimod at the transmitter and provides protection limiting for the STL, which can be flat or preemphasized at 50 or 75 μs. The AGC is turned off in the transmitter-side Optimod.

OPTIMOD-PCn's AGC uses an improved version of the dual-band, window-gated, matrix technology as the AGCs in Orban's 2300, 5300, 8300, 8382, 8400, 8500, 9300, and 9400 Optimods. Therefore, it can effectively substitute for the AGCs in these devices and can help maintain an all-digital signal path throughout the facility. Because OPTIMOD-PCn's AGC is even more advanced than the AGCs in Orban's 2200, 8200, and 9200 Optimods, OPTIMOD-PCn can upgrade the performance of these older products when substituted their AGCs.

Moreover, because OPTIMOD-PCn supports presets that be recalled by remote control, it can be automatically synchronized to the presets on-air at a transmitter-side Optimod when presets are dayparted.

To achieve this match:

A) Recall one of the STUDIO AGC presets. There are three such presets, for flat, 50 µs, and 75 µs preemphasized STLs. See *AGC+LIMIT* on page 3-34.

> You must also configure the peak limiter to match the preemphasis that your STL uses. See step 4.C) on page 2-18.

B) Edit AGC parameters of the OPTIMOD-PCn preset so they are the same as the AGC parameters on-air at the transmitter-side Optimod. Then save your work as an OPTIMOD-PCn User Preset.

> If you use more than one preset in the transmitter-side Optimod, make one OPTIMOD-PCn User preset for each transmitter-side Optimod preset.

C) Adjust the transmitter-side Optimod's input reference level so that the Optimod performs the correct amount of multiband gain reduction (i.e., the same amount of GR that it would have performed if its internal AGC were active).

About OPTIMOD-PCn's Signal Processing Features

Audio Processing Signal Flow

The signal flows through OPTIMOD-PCn in order through the following blocks:

- DC Removal
- Input Conditioning, includes defeatable highpass filtering, lowpass filtering and phase rotation.
- Stereo Synthesizer, auto-detecting and defeatable, uses Orban's classic complementary comb filter technique (stereo processing only)
- Mono Bass processing blends the left and right channels below 80 or 100 Hz (defeatable; stereo processing only).
- Subharmonic Synthesizer creates energy one octave below program energy in the range of 50-90 or 60-120 Hz when such energy is not present at the input and when music is detected.

- Left/Right Phase Skew Correction corrects phase shifts between the left and right stereo channels that could otherwise cause comb filtering in the mono sum (stereo processing only).
- Stereo Enhancement uses upward expansion of the stereo difference signal as triggered by transients in the stereo sum signal.
- Optimix™ dual-mode automatic stereo→surround upmixer (surround processing location)
- Two-Band Gated AGC, with target-zone window gating and silence gating
- Equalization, including high-frequency enhancement
- Downward Expander in five bands
- Multiband Compression in five bands
- Automatic Loudness Control using Orban's third-generation CBS Loudness Controller™ algorithm plus BS.1770 Safety Limiter
- Optimix™ dual-mode automatic stereo→surround upmixer (stereo processing location)
- Peak Limiting of several varieties

Tutorial: More about Dynamics Processing (starting on page 3-28) provides a basic tutorial on the concepts of dynamics processing, including AGC, compression, and peak limiting.

Sample Rate Conversion

The base sample rate of OPTIMOD-PCn's internal processing 48 kHz even when the input and output are configured for 44.1, 96 or 192 kHz; in these cases, synchronous sample rate conversion occurs at the input and output. This 48 kHz rate accommodates a 20 kHz audio bandwidth with a comfortably wide 4 kHz transition band for the anti-aliasing filter. Except for one study that used grossly overspecified anti-alias filters (whose impulse responses were at least four times as long as necessary to achieve flat response to 20 kHz), we are aware of no bias-controlled double-blind studies that have ever suggested that sample rates higher than 48 kHz are audibly superior to 48 kHz (or even that there is any audible difference at all when an optimally-designed anti-alias filter is used).

Moreover, the noise and distortion produced by a given digital filter at 48 kHz is about 6 dB lower than the noise and distortion produced by a filter having the same frequency response but operating at 96 kHz. OPTIMOD-PCn uses many digital filters, both in its equalizer section and for the crossovers in the multiband compressor. Hence, we believe that 48 kHz is the ideal rate for OPTIMOD-PCn's audio processing. Higher sample rates would not only increase CPU usage but would decrease the signal-to-noise ratio of the processing.

Input Conditioning

- A highpass filter with a cutoff frequency of 0.15 Hz removes DC offset from source material without causing significant tilt of low-frequency squarewaves.

- A highpass filter, tunable between 20 Hz and 200 Hz and with selectable 6, 12, 18, or 24 dB/octave slopes, is useful for production applications where it is necessary to remove low frequency rumble from a recording, and in news/talk broadcasting formats.

- A gentle-slope (6, 12, 18, or 24 dB/octave) lowpass filter, tunable between 4 kHz and 15 kHz and with selectable 6, 12, 18, or 24 dB/octave slopes, is useful for production and mastering.

- A defeatable phase rotator makes speech more symmetrical, reducing its peak-to-average ratio by as much as 6 dB without adding nonlinear distortion. Hence, phase rotation can be very useful for loudness processing of speech.

- A very steep lowpass filter can be tuned between 10 kHz and 20 kHz. Set it to complement the bandwidth of the transmission channel that OPTIMOD-PCn is driving. “20 kHz” is a placeholder that is actually “no filter.”

- **Stereo Synthesizer**
- The stereo synthesizer emulates the classic analog Orban 275A automatic stereo synthesizer.

- This process creates an artificial stereo difference signal (L−R) by passing the mono input through a multistage allpass filter. After matrixing with the original mono input (which is the L+R signal) to produce the synthesized left and right channels, the result is a “complementary comb filter” whose notches are spaced in frequency in an approximately logarithmic manner. Because only the L−R signal is created artificially, it cancels out of a mono mixdown, making the synthesizer’s output completely mono-compatible. This processing is only available in stereo mode.

- The synthesizer can be invoked manually or by automatic detection of silence on the right input channel.

- **Mono Bass Processing**
- This applies a steep-slope 80 Hz or 100 Hz highpass filter to the stereo difference signal (L−R). A compensating delay is applied to the L+R signal, making the bandwidth of the transition between mono and well-separated stereo as narrow as possible. This processing is only available in stereo mode.

- We strongly recommend activating mono bass processing when the stereo synthesizer is active.

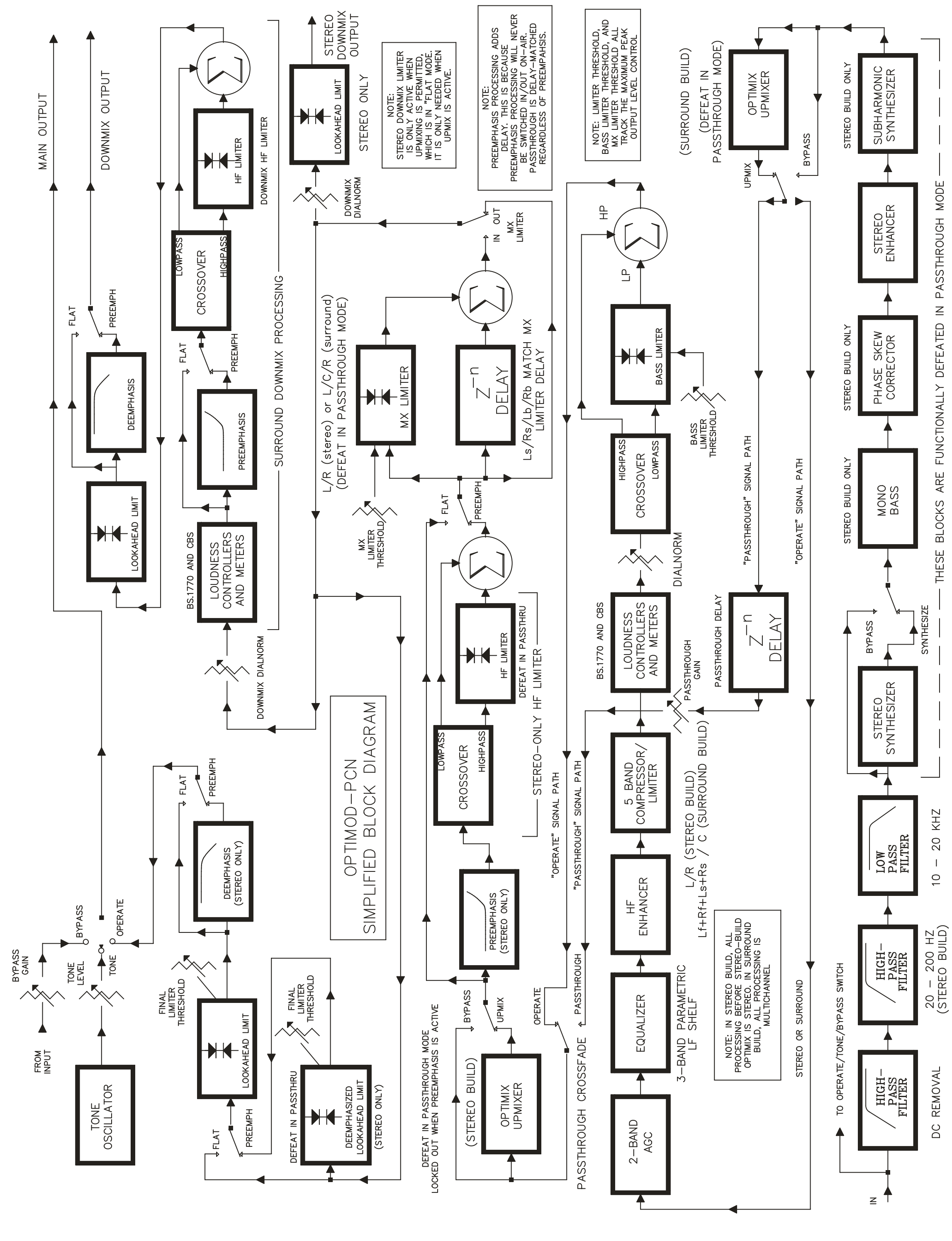


Figure 3-2: OPTIMOD-PCn Digital Signal Processing Simplified Block Diagram

Subharmonic Synthesizer

The subharmonic synthesizer generates subharmonics of fundamental frequencies in the 50-90 or 60-120 Hz range, user-selectable via the SUBHARMONIC CUTOFF FREQ control. The subharmonics are one octave below the frequencies from which they are generated (i.e., 25 to 45 Hz or 30 to 60 Hz) and track the levels of their generating frequencies.

If input program material below 45 or 60 Hz is present, the subharmonic synthesizer automatically reduces the level of the synthesized subharmonics to prevent excess build-up of energy below 45 or 60 Hz.

To prevent introducing unnatural coloration in male speech, the subharmonic synthesizer is defeated when the automatic speech/music detector detects speech. This is particularly critical when the SUBHARMONIC CUTOFF FREQ control is set to 120 HZ because the synthesizer can cause "thumping" sounds behind deep male speech. It is therefore important to keep raw speech in the center of the stereo image, because this is one of the criteria that the speech/music detector uses to discriminate between speech and music.

Stereo Enhancement

OPTIMOD-PCn provides a stereo enhancement algorithm based on Orban's analog 222 Stereo Enhancer, which increases the energy in the stereo difference signal (L−R) whenever a transient is detected in the stereo sum signal (L+R). By operating only on transients, the 222 increases width, brightness, and punch without unnaturally increasing reverb (which is usually predominantly in the L−R channel).

Use stereo enhancement with care if you are driving a low bitrate codec. At low bitrates, these codecs use various parametric techniques for encoding the spatial attributes of the sound field. Stereo enhancement can unnecessarily stress this encoding process.

Because stereo enhancement complements the Optimix 2.0->surround upmixer by making the upmix more dramatic and enveloping, enhancement is available in both stereo and surround modes. In surround mode, the enhancement is coupled to the state of the Optimix upmixer and operates only when Optimix is upmixing.

Gating circuitry detects "mono" material with slight channel or phase imbalances and suppresses enhancement so this built-in imbalance is not exaggerated. It also allows you to set a "width limit" to prevent over-enhancement of material with significant stereo content, and will always limit the ratio of L−R / L+R to unity or less.

Left/Right Phase Skew Correction

The phase skew corrector maximizes the quality of a mono mixdown that might occur in a receiver or player device. At higher frequencies (where audible comb filtering of the mono sum is most likely to occur), the corrector removes phase differences between the left and right channels, converting the HF signal into "intensity stereo" while preserving phase differences at lower frequencies where these differences are important for psychoacoustic "envelopment." The PHASE CORRECTOR CROSSOVER control in the LESS-MORE tab of the active Processing Preset sets the crossover frequency above which phase correction occurs, and IN/OUT activates or defeats the phase corrector via a delay-matched crossfade.

This process can not only correct problems due to phase skew between the left and right channels of an analog recording due to head gap misalignment, it can also correct comb filtering caused by spaced microphones feeding the left and right channels, which can occur on drum kits and other sources that have been multi-miced.

When OPTIMOD-PCn is used as a pre-processor for an FM audio processor like an Orban Optimod-FM, the phase corrector minimizes the amount of energy in the stereo subchannel, which consequently minimizes multipath distortion without compromising stereo separation. It can allow more stereo enhancement to occur for a given amount of multipath distortion. The process also minimizes the amount of peak overshoot during SSB/VSB operation of the stereo encoder (if available), thus minimizing the amount of composite limiting needed to constrain peak modulation to 100%.

Because the phase skew corrector can subtly alter the stereo spatial effect, we recommend using it only as necessary (for example, with formats that play older recordings from the analog era). It can be smoothly activated and defeated via a delay-matched crossfade, so it is practical to do live switching between a preset with the process active and one where it is inactive.

Because it adds considerable delay and uses a significant amount of CPU power, the phase skew corrector can be bypassed completely in I/O > UTILITY. If you are not using it and do not need to activate it smoothly "on-air," bypass it.

This processing is only available in stereo mode and is locked out when the Optimix upmixer is set to AUTO or UPMIX modes. The Optimix phase skew corrector is a somewhat different process that corrects only the center channel of the upmix while leaving the left and right channels unmodified, which maximizes the subjective envelopment of the upmix.

Optimix® Automatic Stereo→Surround Upmixer

The built-in Optimix Stereo→Surround Upmixer is located before the AGC in a surround processing chain. To allow the preceding processing to save CPU cycles by operating in stereo mode, Optimix is located just before the peak limiters in a stereo processing chain.

Optimix can be operated in two modes: FRONTAL and WRAP, determining whether the virtual soundstage is wrapped around the listener (putting the listener "inside the band") or is placed primarily in the front (putting the listener "inside the auditorium").

In surround mode, Optimix's placement in the signal chain means the AGC and multiband compressor process the upmix the same way as they do true surround material. This allows the multiband compressor to correct the frequency and loudness balances of dialog automatically, even if the dialog was sourced from a two-channel stereo mix.

In surround mode, Optimix can automatically sense whether the input material is two-channel stereo or surround. It does this by detecting energy in the center, left surround, and right surround input channels. In automatic switching mode, if Optimix detects energy in any of these channels, it will immediately bypass the upmixer. If there is no energy in any of these channels, Optimix will create a realistic-sounding surround mix from stereo input.

In stereo mode, Optimix is located after the loudness controllers in the signal path. No automatic switching is provided because its input is always stereo.

Peak limiters follow Optimix and operate in five-channel mode. To save CPU, in MX mode only the Lf, Rf, and C peak limiters use the MX algorithm. Because the peak level of the upmixed L and R surround channels is typically much lower, these channels require little or no peak limiting and can use the non-MX look-ahead limiting algorithm without adding artifacts.

Optimix includes a sophisticated delay/phase correction algorithm that prevents phase cancellations between the left and right channels from affecting the crispness and high frequency response of the upmixed center channel. It does not affect other channels in the upmix. This is particularly important with older program material that may have been archived on analog tape, which is frequently subject to phase skew between the left and right channels.

See *Optimix™ Upmixer Controls* on page 3-45.

Two-Band Gated AGC

The AGC is a two-band device, using Orban’s patented “master/bass” band coupling. It uses a linear-phase crossover.

In surround mode, the total power in the input channels (RMS) determines the gain reduction, so all channels receive equal amounts of gain reduction.

In stereo mode, there are two gain-control sidechains, one for each stereo channel. To preserve RMS operation, the MAXDELTAGR control operates a constant-power, symmetrical panpot. When the MAXDELTAGR control is set to 0, it applies equal amounts of left and right energy to both sidechains, which causes the left and right sidechains to track each other, maintaining 100% stereo coupling. As the MAXDELTAGR control is set to higher values, it progressively applies less right-channel energy to the left-channel sidechain and vice-versa. Setting the control to a higher value will partially correct stereo program material with left/right channel imbalances. Higher settings should be used with caution, as they can cause instability in the stereo image.

The AGC contains a compression ratio control that allows you to vary to ratio between 2:1 and essentially ∞:1. Lower ratios can make gain riding subtler on critical formats like classical and jazz. See Figure 3-4.

The AGC incorporates target-zone gating. If the input program material's level falls within a user-settable window (typically 3dB), the release time slows to a user-determined level. It can be slow enough (0.5 dB/second) to effectively freeze the operation of the AGC. This prevents the AGC from applying additional, audible gain riding to material that is already well controlled. It also lets you run the AGC with fast release times without adding excessive density to material that is already dense. Figure 3-5 shows the behavior of the window for AGC DRIVE = 12 dB.

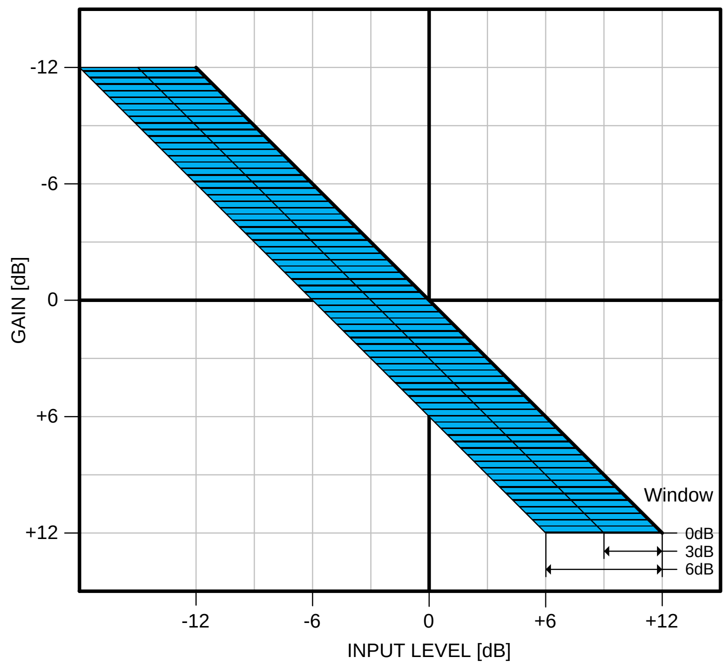


Figure 3-3: AGC Gain vs. Input Level Showing Window

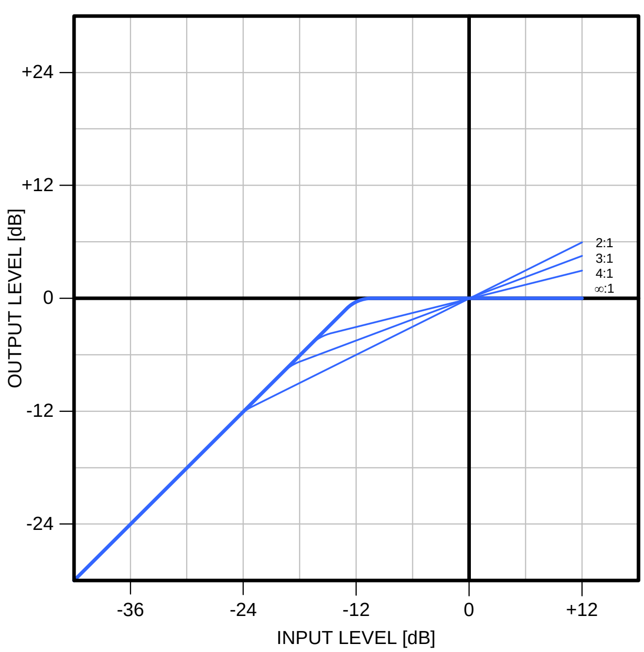


Figure 3-4: AGC Output vs. Input as a Function of the AGC Ratio Control

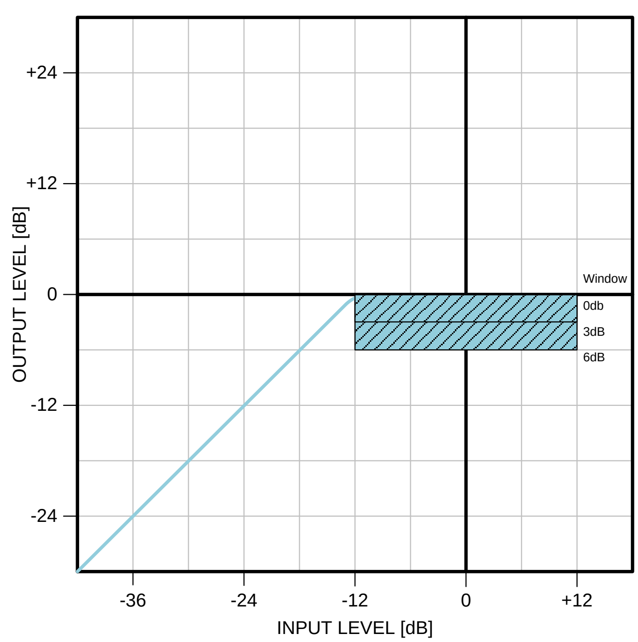


Figure 3-5: AGC Output vs. Input Showing Window

The AGC has a dedicated silence-gating detector whose threshold can be set independently of the silence gating applied to the multiband compressor. In stereo mode, the gating detector is operated from the output of the left-channel panpot implemented by the MAXDELTAGR control, so if this control is set for fully uncoupled operation (not recommended) and there is energy only in the right channel, then the gate will activate.

Equalization

OPTIMOD-PCn has a bass shelving equalizer, four bands of fully parametric bell-shaped EQ, and a dynamic high-frequency enhancer.

You can set the slope of the bass shelving EQ to 6, 12, or 18 dB/octave, adjust the shelving frequency and set the amount of equalization.

OPTIMOD-PCn's bass, midrange, and high frequency parametric equalizers have curves that were modeled on the curves of Orban's classic analog parametrics (like the 622B), using a sophisticated, proprietary optimization program. The curves are matched to better than 0.15dB. This means that their sound is very close to the sound of an Orban analog parametric. They use very high quality filter algorithms to

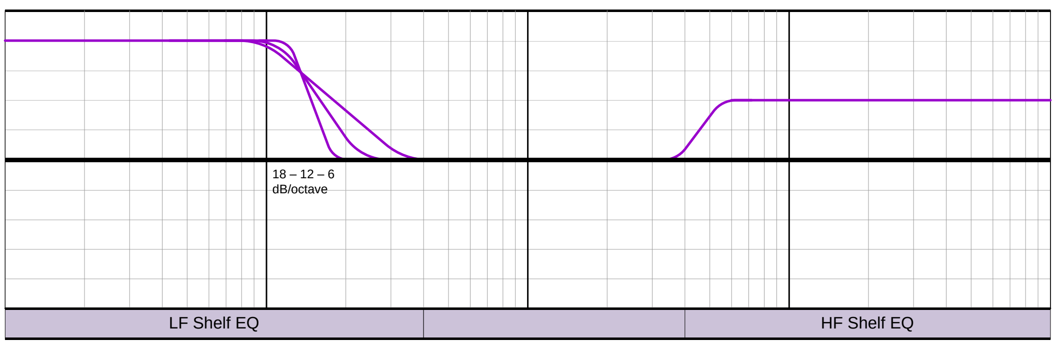


Figure 3-6: LF and Dynamic HF Shelving Equalizer Frequency Response

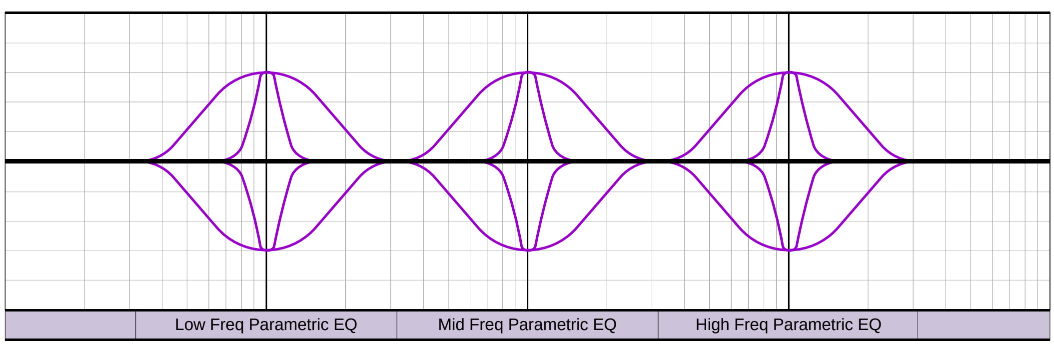


Figure 3-7: Parametric Equalizer Frequency Response for Narrow and Wide Bandwidths

ensure low noise and distortion.

The HF Enhancer is a program-controlled HF shelving equalizer that intelligently and continuously analyzes the ratio between broadband and HF energy in the input program material. It can equalize excessively dull material without over-enhancing bright material. It interacts synergistically with the five-band compressor to produce sound that is bright and present without being excessively shrill.

The BRILLIANCE control adds additional gain before the Band 5 multiband compressor, so it acts like a steep-slope shelving equalizer.

See Figure 3-6 and Figure 3-7 on page 3-18 for illustrative frequency responses of the shelving and parametric equalizers.

Multiband Compressor/Limiter

The multiband compressor/limiter operates in five frequency bands. Each band compressor has a KNEE and RATIO control. A soft knee and gentle ratio are particularly useful in production and mastering applications, allowing subtle compression that retains as much of the dynamics of the input program material as the operator desires. (The SOFT KNEE AND SOFT KNEE MX presets are intended for mastering; they exploit the soft knee features.) These features are also exploited in the TV 5B DRAMA preset.

Several band-coupling controls allow the gain reduction of a given band's compressor to be affected by the gain reduction in its neighboring band's compressor. These coupling controls allow anything from quasi-wideband compression to fully independent multiband compression.

The **surround multiband compressor** (available only in Surround versions of OPTIMOD-PCn) is always stereo-coupled and has two main sidechains.

- *The first surround sidechain responds to the power (RMS) sum of all channels. However, to prevent sibilance in the center channel from ducking high frequencies in the remaining channels, the center channel does not affect the gain reduction in bands 4 and 5.*

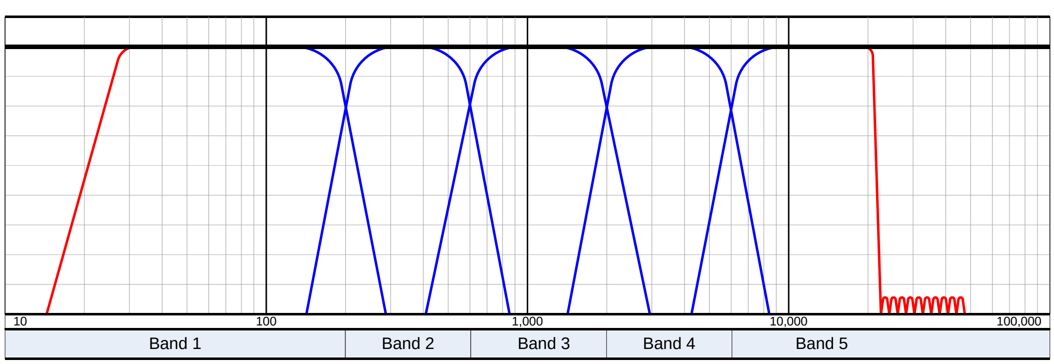


Figure 3-8: Multiband Compressor Crossover Filters (Illustrative)

As in the AGC, the SURROUND OPTIMIZATION control for the active preset determines how the rear channels are weighted in the RMS sum.

This sidechain applies the slow “compression” component of the gain reduction identically in all channels except for the center. However, each channel also has an individual “limiting” function that is not stereo coupled. This prevents fast transients in one channel from audibly modulating the gains of the other channels. Because this “limiting” function operates quickly, it does not affect stereo imaging.

> Do not confuse this “limiting” function with the 1600’s look-ahead limiter, which is located after the Loudness Controller in a given processing chain and which has much faster attack and release times.

- *The second surround sidechain responds to the center channel alone. It provides both a “compression” and “limiting” function.*

The dual-sidechain architecture’s main application is to improve dialog intelligibility by preventing strong energy in the non-center channels from drowning out dialog in the center channel even when the original mix was miscalculated. To achieve this, the center channel’s gain sometimes needs to be increased with respect to the other channels and the dialog needs to be re-equalized automatically by multiband compression. Moreover, this dual-sidechain architecture permits the multiband compressor to de-ess center channel dialog as necessary without affecting the other channels.

Completely independent compression of the dialog channel can cause unnatural-sounding level imbalances between the dialog and the remaining program elements. To address this, there are two coupling controls for each frequency band in the multiband compressor. These coupling controls work by constraining the difference (in dB) between the center channel’s gain reduction and the main channel’s gain reduction. To do this, they clamp the center channel’s gain reduction with respect to the main channel but do not affect the gain reduction in the main channel. In other words, the center channel’s gain reduction is either the output of the center channel’s compression sidechain or a fixed offset from the main channel’s gain reduction. There are separate controls to set the maximum positive and negative offsets. The constraints are applied to the entire center gain reduction signal (compression plus limiting).

To preserve intelligibility, it is important to prevent the center channel’s gain reduction from becoming too large with respect to the other channels. The BX MAIN>CENTER MAX +DELTA GR control sets the maximum positive gain reduction difference in each band. When the five-band structure is used, it is common to set this control to 0 dB in Bands 1-3. To facilitate de-essing (and because Bands 4 and 5 in the center channel do not contribute to the gain reduction in the main sidechain), we recommend setting the B4 and B5 MAIN>CENTER MAX +DELTA GR controls to allow some extra gain reduction to occur in center channel Bands 4 and 5. It is OK to set this control to OFF in Band 5, which allows unconstrained de-essing above 6 kHz. The TVxxxx factory presets provide examples.

Meanwhile, a BX MAIN>CENTER MAX −DELTA GR sets the maximum negative gain reduction difference in each band, which constrains the amount of dynamic dialog boost. It also constrains the degree to which the center channel compressor can pull up noise. If there are program elements other than dialog panned into the center channel, the control constrains the amount by which the front stereo image can be narrowed. 3 dB is a typical setting.

Each band in the **stereo multiband compressor** (provided in stereo versions of OPTIMOD-PCn) has two sidechains—one for the left and one for the right channel. You can separately set the left/right coupling of each band anywhere from 100% stereo coupled to fully independent.

Although the multiband compressor's architecture has two sidechains, OPTIMOD-PCn does not allow two different programs to be routed through a stereo Processor. If you need one or more channels of mono processing, use an instance of mono processing for each. This allows each Processor to have separate setup and processing controls applied to it.

- In STEREO mode, there is one LOUDNESS LEVEL meter, AGC GATE indicator, and MB GATE indicator. However, you can still uncouple each band in both the AGC and multiband compressors to a variable extent—anywhere from perfect stereo coupling to completely uncoupled operation. The coupling control determines the maximum amount of gain difference permitted between the left and right channels in a given band and therefore the amount of stereo image shift permitted in each frequency band.

Parallel Compression

The parallel compression mode drives downstream processing with the sum of the input and output of the multiband compressor. As compressor gain reduction increases, the compressor contributes less and less to the overall output. Parallel mode is mainly useful for classical music, preserving the dynamic impact of loud material while making quiet material more audible. Typical applications are preparing material for in-flight entertainment or other noisy environments.

There are two specific controls for parallel mode (in the MULTIBAND tab): COMPRESSOR MODE [Inline, Parallel] and COMPRESSION THRESH OFFSET [0 to –60 dB, 1 dB steps]. Additionally, in parallel mode the MULTIBAND DRIVE control is repurposed so that it sets the amount of compressor output that is added to the compressor's input. Its setting indicates the amount of amplification of quiet material that occurs when the compressor produces no gain reduction. It does not affect the amount of compressor gain reduction.

The COMPRESSION THRESH OFFSET control is only active in Parallel mode. It moves all compressor thresholds in the five-band compressor down by the same amount. It is needed because the MULTIBAND DRIVE control does not affect the amount of compressor gain reduction and because the individual bands' COMPRESSION THRESHOLD controls do not have enough range to achieve the desired amount of gain reduction with all program material.

Parallel compression is usually used to bring up quiet material while minimally affecting loud material. Hence, the compressor threshold must be much lower than normal so that the compressor develops maximum gain reduction with high-level material. Adjust the COMPRESSION THRESH OFFSET control so that the compressor gain reduction meters are active but on-scale during quiet material that you wish to make louder. Typical settings are –30 to –40 dB.

It is normal for the five-band compressor gain reduction meters to indicate full-scale during loud material. (The maximum gain reduction is clamped internally to 40 dB.)

The COMPRESSION THRESH OFFSET control also scales the MB GATE THRESHOLD and the DOWN EXPANDER thresholds downward so that these features can be effective with source material that has low intrinsic noise, like most digital recordings. These features can help resist increasing audible noise (such as air conditioning rumble) during the quietest parts of the program.

Note that parallel compression can be effective and unobtrusive even with faster settings of the MULTIBAND RELEASE control.

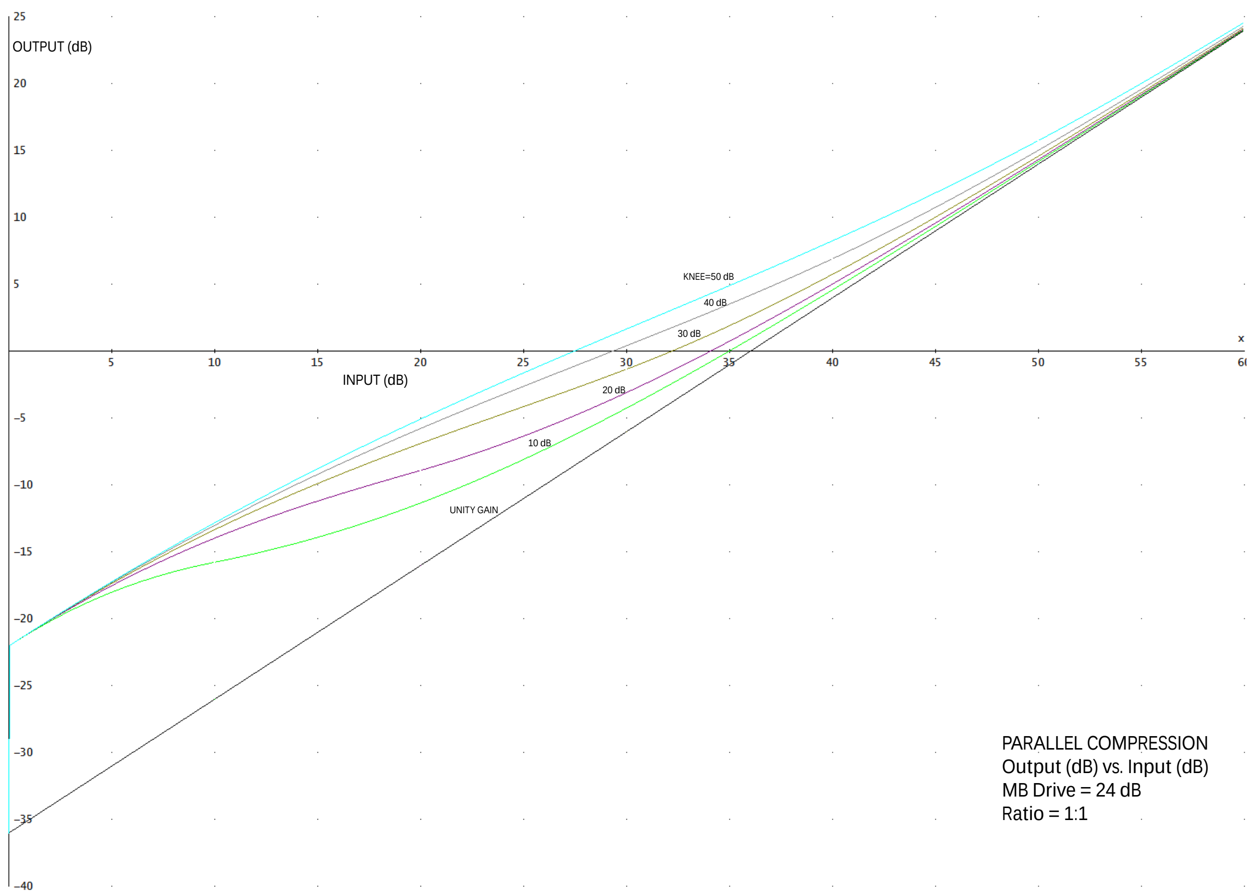


Figure 3-9: Parallel Compression Input vs. Output (dB)

To prevent phase cancellation between the compressor's output and input, the summation occurs immediately after the compressor's VCAs, and the five-band crossover is still in the signal path. ALLPASS crossover mode produces phase rotation, so LINEAR crossover mode is most appropriate for the kind of program material typically processed with parallel compression.

LFE Processing

In a surround Processor, the LFE channel has its own dedicated compressor with ATTACK TIME and THRESHOLD controls. Its other settings (such as RATIO and BREAKPOINT) track the settings of the B1 compressor. To constrain build-up of LFE energy, the B1>LFE COUPLING control clamps the maximum gain of the LFE channel with respect to the band 1 compressor in five-band mode and the Bass band in two-band mode. For example, when this control is set to 3 dB (the default) and the B1 compressor exhibits 12 dB of gain reduction, the LFE channel can never have less than 9 dB of gain reduction even if the LFE compressor would have produced less than 9 dB of gain reduction if uncoupled.

Peak Limiter

OPTIMOD-PCn's look-ahead true-peak limiter prevents overshoots by examining a few milliseconds of the unprocessed sound before it is limited. This way the limiter can anticipate peaks that are coming up. The limiter's sidechain is oversampled to 384 kHz to prevent significant overshoot from occurring after sample rate conversion or D/A conversion. In stereo and mono instances additional pre-processing, borrowed from MX SOFT mode (see below), reduces potential artifacts caused by wide-band limiting. Surround instances do not use the additional pre-processing because the target loudness of surround instances is usually –23 or –24 LUFS, which requires little or no peak limiting.

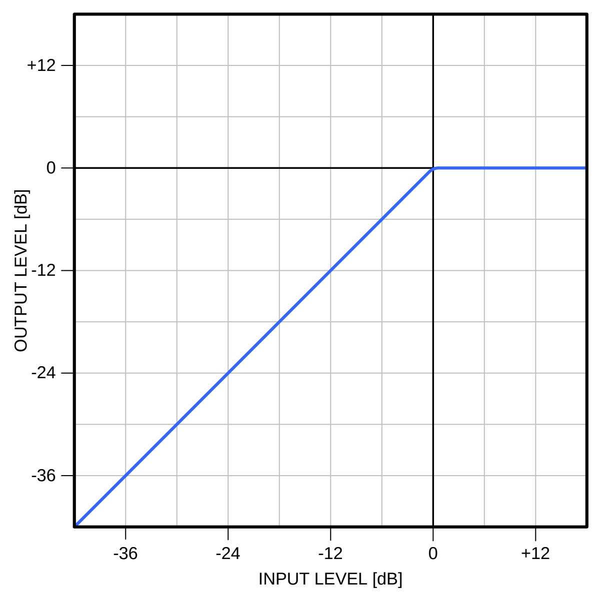


Figure 3-10: Peak Limiter Output vs. Input

The limiter's drive level mainly determines the gain reduction. Two cascaded gain controls set this drive level. One is MB LIMIT DRIVE; the second is a "hidden" control whose gain is set by the active TARGET LOUDNESS value. Additionally, the 100% OUTPUT LEVEL control is a limiter threshold control that increases the amount of limiting as the control is turned down (see *Figure 1-4* on page 1-21).

For more about the concept of target loudness, see *About Target Loudness and ITU-R BS.1770* starting on page 1-16.

You can activate MX peak limiter technology, which introduces fewer audible artifacts than pure look-ahead limiting when a large amount of limiter gain reduction is required and can add some "exciter"-like harmonic enhancement, depending on the setting of the HF DISTORTION CONTROL control. MX limiting is a CPU-intensive process that can add up to 240 ms of delay and is mainly beneficial if the target loudness is above −12 LUFS. (When the target loudness is below −12 LUFS, the peak limiter usually operates lightly.)

The MX limiter has two overshoot compensator modes: HARD and SOFT. HARD is very CPU-intensive and is mainly useful when you need to create very high loudness, such as that found in some "hypercompressed" CDs. Additionally, SOFT CAN be cleaner sounding at the expense of producing some audible gain pumping. The choice of HARD or SOFT modes must be made by ear.

HARD mode is only available when the preemphasis is FLAT.

In surround mode, or if you are in stereo mode and upmixing via Optimix, MX limiting is applied to the left, center, and right channels only. The surround channels, which normally have lower peak levels, are limited using the non-MX limiter. This provides the advantages of MX limiting without excessively increasing CPU usage.

In stereo mode and with Optimix defeated in I/O Setup, only the left and right MX limiters are active, saving CPU cycles.

Loudness Control

OPTIMOD-PCn's third-generation CBS Loudness Controller and BS.1770 Safety Limiter cooperatively and automatically control subjective loudness.

Useful in sound-for-picture applications, particularly when government regulations like the CALM Act require control of subjective loudness, the third-generation CBS Loudness Controller follows the multiband compressor in the signal flow diagram. This placement reduces the drive level into the peak limiter when the loudness controller produces gain reduction. This minimizes peak-limiter-induced artifacts.

The loudness controller takes into account control settings that affect the peak limiter so that the loudness controller approximately monitors the loudness at the limiter's output, not the loudness at the multiband compressor's output. It does so by scaling its sidechain drive by two gain factors: (1) the MB LIMIT DRIVE control and (2) the 1600's active TARGET LOUDNESS value, which is determined either globally or by the active preset depending on the active preset's TARGET LOUDNESS setting. (See *Figure 1-4* on page 1-21 and *Peak Limiter* on page 3-23.) The TARGET LOUDNESS value is the same as value of "dialnorm." OPTIMOD-PCn contains a BS.1770 "integrated" meter with an integration time of 10 seconds. To comply with these standards, TARGET LOUDNESS is customarily set to −24 in the United States and −23 in the EU.

The Loudness Controller constrains the loudness of most commercials well enough to eliminate viewer annoyance. It works by constantly monitoring the subjective loudness of the 1600's output. The subjective loudness is a single value that represents the listener's impression of the loudness in the listening room. It takes into account the contribution of all stereo and surround channels.

When subjective loudness would otherwise exceed the threshold set by the LOUDNESS THRESHOLD control, the CBS Loudness Controller reduces the gain of material above 200 Hz, preventing loudness from exceeding the threshold. To prevent the loudness controller from causing too much dynamic bass boost, you can use the

LOUDNESS CONTROLLER BASS COUPLE control to limit the maximum difference between the gain of the band below 200 Hz and the band above 200Hz. For example, when this control is set to 3 dB and the loudness controller's gain reduction is 10 dB, the gain reduction below 200 Hz will be 7 dB. However, if the loudness controller's gain reduction is 2 dB, the gain reduction below 200 Hz will be 0 dB because the difference is now less than 3 dB.

The loudness controller is triggered mainly by program material that has a lot of energy between 1 and 7 kHz, which is the ear's most sensitive range. The five-band compressor can automatically re-equalize such program material so that it does not induce unnatural-sounding gain reduction in the loudness controller, and will de-ess extremely sibilant program material before the loudness controller receives it.

The loudness controller's attack and release times are tuned to match the loudness integration times of the ear and are program-adaptive. Only the attack time is user-adjustable.

If you feel that the Loudness Controller is not controlling the loudness of commercials or other subjectively loud program material sufficiently well, you may wish to set the threshold lower, forcing the Loudness Controller to do more work. You may also wish to activate the BS.1770 Safety Limiter.

In the LOUDNESS GAIN REDUCTION meter, the gain reduction of the BS.1770 safety limiter (in dB units) is stacked with gain reduction of the CBS loudness controller. The CBS controller gain reduction appears in blue, while the BS.1770 safety limiter gain reduction appears in cyan. The CBS Loudness Controller produces both fast and slow loudness control; the fast control rides on top of the slow control. You can easily see this dual-speed operation on the LOUDNESS GR meter. The LOUDNESS ATTACK control determines how much fast control the Loudness Controller produces. As the control is turned down toward 0%, it allows longer and longer loudness peaks to pass through.

> Because of the system topology, the LOUDNESS LEVEL (CBS) meter assumes that the LOUDNESS ATTACK control is always set to 100% and does not indicate the effect of lower settings. As long as the control is set to 50% or higher, this limitation should not have any significant effect on the loudness level meter's accuracy, and regardless of setting does not affect the BS.1770 loudness meter's accuracy. However, it is wise to double-check the effect of the LOUDNESS ATTACK control on subjective loudness by listening tests and/or use of an external loudness level meter like the free Orban Loudness Meter for Windows, which uses the same algorithm as the built-in LOUDNESS LEVEL meter (www.orban.com/meter).
>
> Moreover, because of their placement in the signal path the CBS loudness controller and meter are unaware of the amount of gain reduction that occurs in the peak limiter. In normal sound-for-picture applications (with target loudness around −23 LUFS), the limiter operates very lightly (if at all) and does not affect the loudness, so loudness control and metering are very accurate. If the target loudness is high (for example, −8 LUFS, which is a typical Target Loudness value in the radio-style presets), the peak limiter may be active enough to reduce loudness by 1 or 2 LU, and this will cause the LOUDNESS LEVEL meter to over-indicate the loudness by this much. (Again, you can use the free Orban Loudness Meter to check

the actual loudness at OPTIMOD-PCn's output, and this issue does not affect the built-in BS.1770 meter.) Note that all target loudness values recommended in EBU R-128 are low enough to cause the loudness meter and controller to be very accurate.

The Loudness Controller may reduce the dramatic effect of certain sounds in entertainment programming, like gunshots, explosions, or screeching tires. Operators may therefore want to turn the Loudness Controller on during commercial breaks and off during normal programming. All sound-for-picture presets have the Loudness Controller on. The easiest way to handle this situation is to start with your preferred preset, turn the Loudness Controller off, and then save the result as a User Preset. Using one of the 1600's remote control mechanisms, recall the "with loudness controller" and "without loudness controller" presets as desired.

Turning down the LOUDNESS ATTACK control provides another way to maintain the dramatic impact of loudness transients in dramatic programming; it can let gunshots and the like through while still constraining long-term loudness to a fixed threshold. While this is an easy solution that does not require your automation system to tell the 1600 when to recall presets, it is not ideal because there are some short-term loudness events, like sibilance, applause, and whistles, that can be annoying to audiences. You can use the Speech-Mode B4 and B5 compressor threshold controls to accomplish the same goal.

Another loudness control strategy is this: Instead of using two presets with and without loudness control (as described above), you can create presets with different settings of the LOUDNESS ATTACK control (and possibly different settings of the LOUDNESS THRESHOLD control as well). Try a slow attack (50% or below) for dramatic programming and a faster attack (70%) for commercial breaks. This will maintain some automatic loudness control for dramatic programming while controlling the loudness of commercial breaks more rigorously.

Note that the Loudness Controller operates with reference to an absolute subjective loudness threshold that does not adapt to program context. This means that if there is a transition between very quiet program material (like footfalls through rustling leaves) and a commercial, the commercial may *still* seem offensively loud even though the Loudness Controller is controlling its loudness correctly with reference to other sounds that reach full-scale loudness. Philosophically, this is inevitable; the Loudness Controller cannot reduce the level of the commercial to the level of rustling leaves without destroying the effectiveness of the commercial and angering the sponsor!

BS.1770 Safety Limiter

Following the CBS Loudness Controller is a BS.1770 Safety Limiter that will prevent a BS.1770-2 (or higher) loudness meter with 10-second integration time from indicating higher than the setting of the BS.1770 THRESHOLD control, which is found in the MULTIBAND page. The BS.1770 THRESHOLD control is part of the on-air processing preset, so if you change the setting of this control from its default value (which we recommend doing), you should save your work as a User Preset.

Because some organizations will disqualify an automatic loudness controller if it causes a BS.1770 meter to read higher than a specified threshold, all of the "TV" factory presets have this controller active with the BS.1770 THRESHOLD control set to 0 LU. If your organization does not have a strict policy about processing for the BS.1770 meter, we recommend that you edit your preferred preset by setting this control anywhere from +2 to OFF and then saving the result as a User Preset.

The BS.1770 safety limiter receives the output of the CBS Loudness Controller and drives the final peak limiters. The total amount of loudness control-induced gain reduction is the sum of the gain reduction produced by the CBS Loudness Controller and the gain reduction produced by the BS.1770 safety limiter. The gain reduction produced by the BS.1770 safety limiter changes slowly, seldom exceeds 2 dB, and is indicated by the cyan section of the LOUDNESS GR meter. The gain reduction produced by the CBS Loudness Controller may change slowly or quickly (depending on the nature of the program material), appears in blue, and rides on top of the BS.1770 gain reduction. The peak reading of the meter thus shows the total gain reduction that both controllers produce.

We included this safety limiter for customers whose policies require the BS.1770 loudness meter reading to be constrained below specified threshold regardless of how loud human listeners perceive the program to be. Our experience suggests that the BS.1770-2 meter will often over-read material with unusually low peak-to-average ratios, like highly produced commercials and promos. Strict reliance on the BS.1770 meter can therefore make such material sound unnaturally quiet compared to surrounding material, so we prefer the sound when the CBS Loudness Controller is used exclusively for loudness control.

For a more detailed discussion of these issues, refer to Appendix A: Using the ITU BS.1770 and CBS Loudness Meters to Measure Loudness Controller Performance, starting on page 3-93.

Downmix Processing

OPTIMOD-PCn includes a dedicated downmix processor that applies correct loudness control and peak limiting to the stereo downmix and makes the downmix available as a separate output. Two outputs are available and each can have a separate target loudness applied to it.

In surround mode, the downmix chains include loudness control that both outputs share. Each output has a dedicated high frequency limiter and a look-ahead (non-MX) peak limiter.

The difference in Optimix's placement between stereo and surround modes allows the downmix processing to be simplified in stereo mode. Because Optimix's output mixes down to the original stereo, the loudness control originally applied to the main stereo signal, pre-Optimix, also works for the downmix, so the downmix processor only needs to do high frequency limiting and peak limiting.

The high frequency limiters only operate when the downmix operates preemphasized, as set in I/O > DOWNMIX OUTPUT. When the downmix processor is operated preemphasized, it can correctly control the modulation of an analog television or FM transmitter. Although it cannot achieve "competitive loudness" in FM broadcast, the downmix processor can provide excellent performance in analog television applications, and in digital television applications where the target loudness is below –20 LUFS.

Tutorial: More about Dynamics Processing

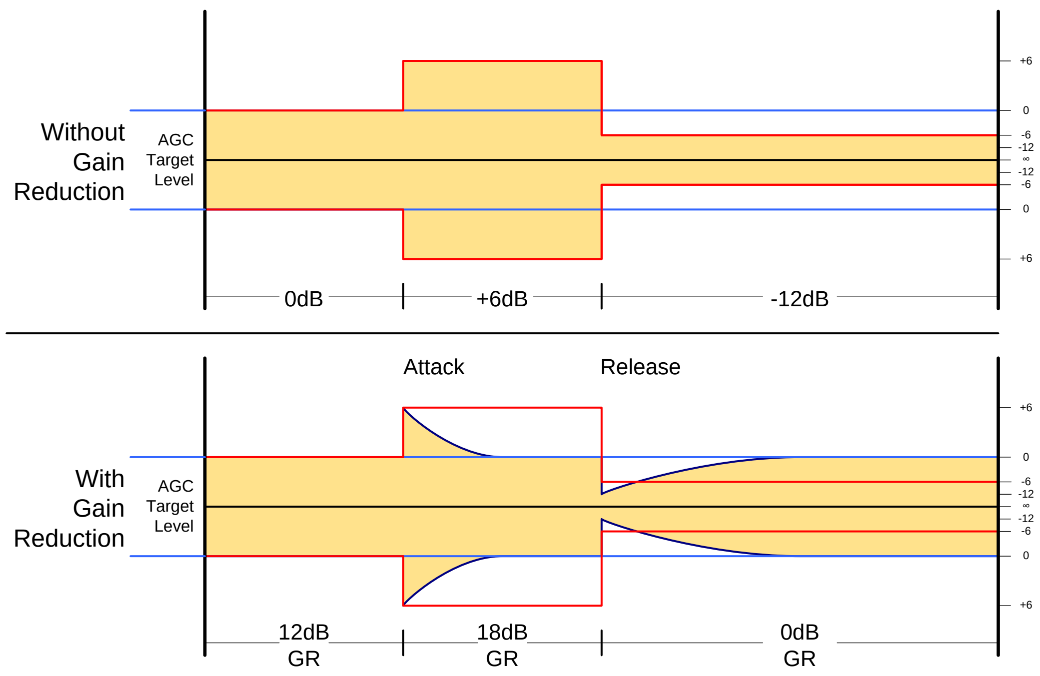


Figure 3-11: AGC Attack and Release

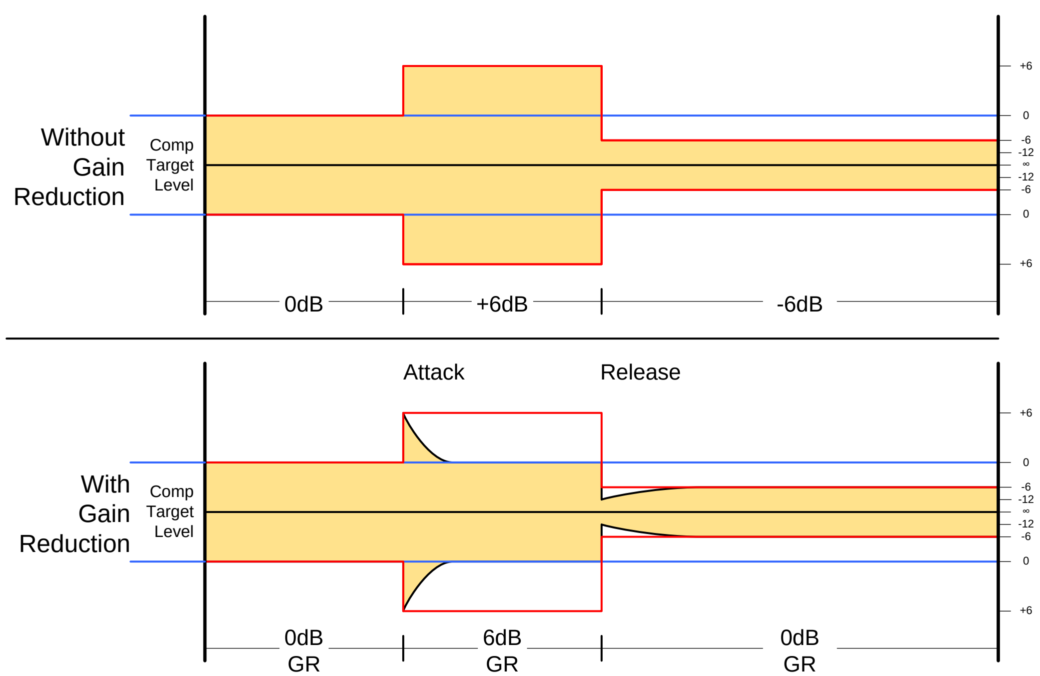


Figure 3-12: Compressor Attack and Release

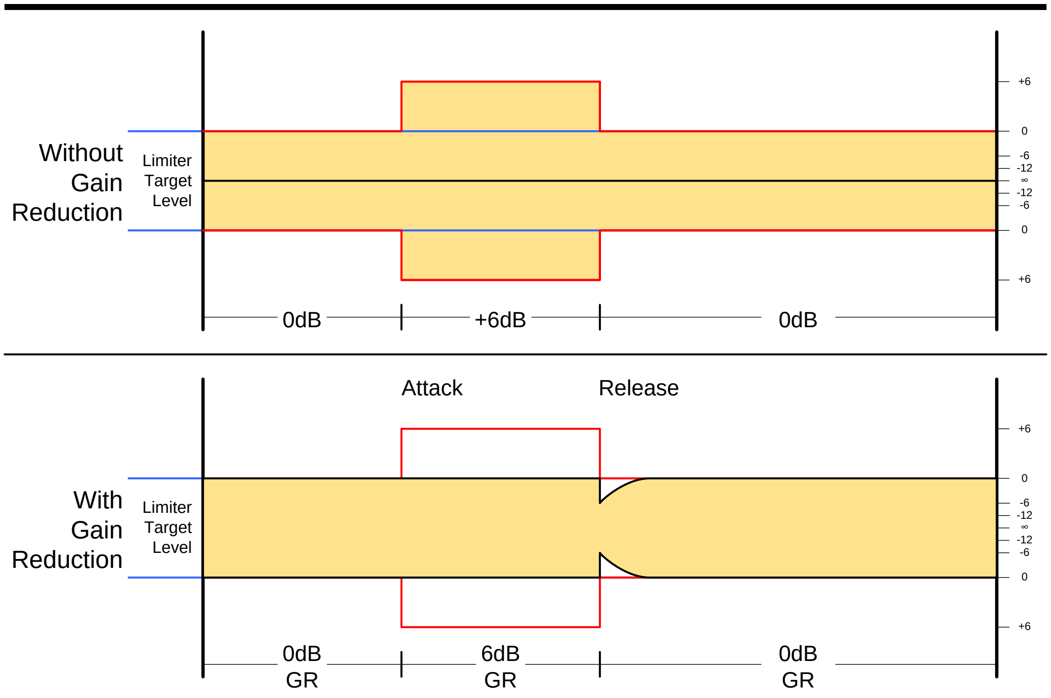


Figure 3-13: Peak Limiter Attack and Release

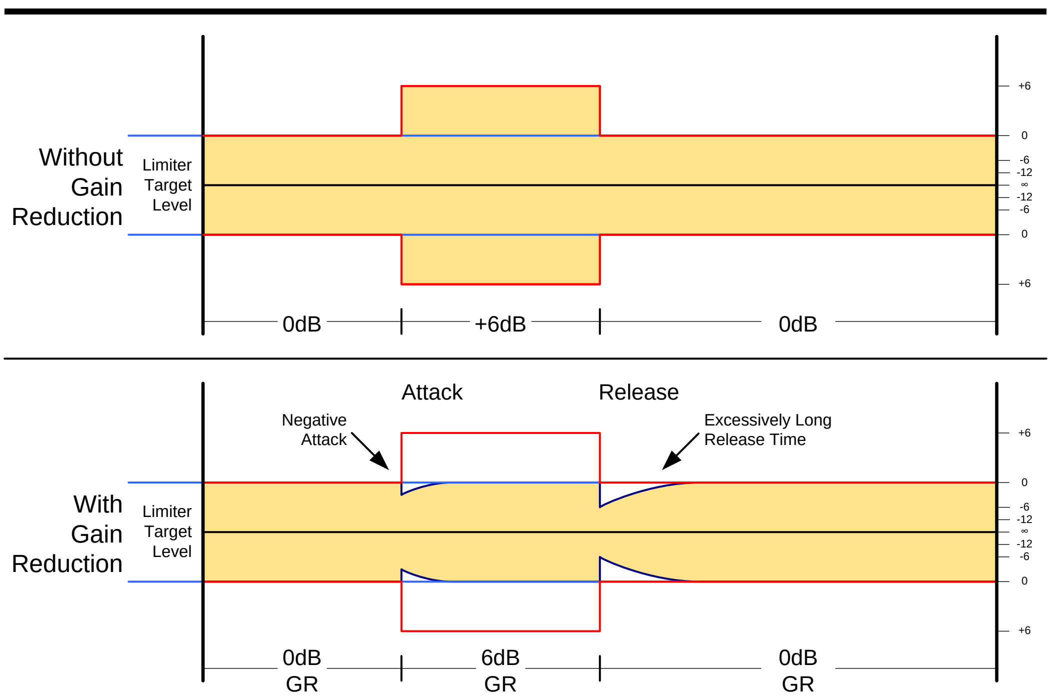


Figure 3-14: Compressor Attack and Release (Bad Design)

OPTIMOD-PCn's AGC, multiband compressor, and peak limiter are all forms of *dynamics processing.* Their main differences are the speed of the attack and release times and their compression ratios. In OPTIMOD-PCn's AGC and multiband compressor, these parameters are all user-adjustable. To prevent peak overshoots, the peak limiter must have an instantaneous attack time and the infinite compression ratio.

Figure 3-12, Figure 3-13 and Figure 3-14 show the audio level as a function of time and are conceptual and simplified. In fact, the attack and release time characteristics of OPTIMOD-PCn's AGC and multiband compressors automatically adapt themselves to the program material they are processing.

Figure 3-11 shows the slow attack and release characteristics typical of AGC. This diagram shows an infinite compression ratio; lower ratios are available in OPTIMOD-PCn's AGC. The AGC performs slow gain riding; it does not increase program density.

Figure 3-12 shows the attack and release times found in OPTIMOD-PCn's multiband compressors, which are substantially faster than the AGC. The multiband compressor is designed to be able to increase program density if the user wants, but also has slow attack and release times available if the user wants to retain the density of the original program material.

Figure 3-13 shows a typical peak limiter attack and release characteristic. The attack is instantaneous to prevent overshoots. Figure 3-13 is not drawn to the same time scale as Figure 3-11 and Figure 3-12; the release time of OPTIMOD-PCn's peak limiter is hundreds of times faster than that of the AGC or multiband compressor.

Figure 3-14 shows one of the many things that can go wrong in a badly designed compressor. The attack causes more gain reduction than necessary to reduce the output level to the threshold of compression, and the gain must then recover to bring the output level to the compression threshold. This attack overshoot and subsequent release causes audible "gulping" or "gain pumping," which gives the impression that the program level is constantly changing in an audibly obtrusive way. The gain pumping exacerbated by a release time that is too slow, so that the audio level starts out too quiet at the beginning of the release period and then slowly and obviously recovers to its final value. While this kind of sound is sometimes used for artistic purposes in production and mastering, in a processor like OPTIMOD-PCn (which is designed to handle all types of program material gracefully without introducing objectionable side-effects), it imposes an unstable sonic signature on all program material and is considered one of the marks of incompetently-designed processing.

Customizing OPTIMOD-PCn's Sound

See *Figure 3-2: OPTIMOD-PCn Digital Signal Processing Simplified* Block Diagram on page 3-13.

The subjective setup controls on OPTIMOD-PCn give you the flexibility to customize your station's sound. Nevertheless, as with any audio processing system, proper adjustment of these controls consists of balancing the trade-offs between loudness, density, and audible distortion. The following pages provide the information you need to adjust OPTIMOD-PCn controls to suit your format, taste, and competitive situation.

When you start with one of our Factory presets, there are two levels of subjective adjustment available to you to let you customize the Factory preset to your requirements: Basic Control and Advanced Control.

Basic Control

The single LESS-MORE control changes many different subjective setup control settings simultaneously according to a table that we have created in OPTIMOD-PCn's factory presets, which are files with the Windows read-only attribute. In this table are sets of subjective setup control settings that provide, in our opinion, the most favorable trade-off between loudness, density, and audible distortion for a given amount of processing. We believe that most OPTIMOD-PCn users will never need to go beyond the LESS-MORE level of control, because the combinations of subjective setup control settings produced by this control have been optimized by our audio processing experts on the basis of years of experience designing audio processing, and upon hundred of hours of listening tests.

Unlike previous Optimods, the LESS-MORE control has been designed keep loudness constant when adjusted. Instead, turning it up typically changes texture by increasing the multiband compressor drive (and this the amount of multiband compressor gain reduction), while decreasing the peak limiter drive appropriately.

To change a preset's loudness, change the value of TARGET LOUDNESS, which you can do by editing the local value of TARGET LOUDNESS in the processing preset (LESS-MORE tab) or setting that value to GLOBAL and then changing the global TARGET LOUDNESS value (in I/O MIXER > GLOBAL). Changing the value of TARGET LOUDNESS will change the drive level into the peak limiter (and hence, will change its gain reduction). This does not change the indication of the loudness meter because it indicates loudness relative to the active TARGET LOUDNESS value.

In the "video" presets, the LESS-MORE control also sets the average amount of dynamic range control provided by the processing. As you go from less to more, the loudness of loud sounds will stay about the same but the loudness of quieter sounds will increase. Because of OPTIMOD-PCn's sophisticated gating circuits, very quiet material like background sounds, quiet underscoring, hiss, and hum will not be pumped up.

You need not (in fact, cannot) create a sound entirely from scratch. All User Presets are created by modifying Factory Presets or by further modifying Factory Presets that have been modified previously. It is wise to set the LESS-MORE control to achieve a sound as close as possible to your desired sound before you make further modifications at the Advanced Modify level. This is because the LESS-MORE control gets you close to an optimum trade-off between loudness and artifacts, so any changes you make are likely to be smaller and to require resetting fewer controls.

In OPTIMOD-PCn, LESS-MORE affects only the dynamics processing (compression and peak limiting)—OPTIMOD-PCn's equalization and stereo enhancement are decoupled from LESS-MORE. You can therefore change EQ or stereo enhancement and not lose the ability to use LESS-MORE. When you create a user preset, OPTIMOD-PCn will automatically save your EQ and stereo enhancement settings along with your LESS-MORE setting. When you recall the user preset, you will still be able to edit your LESS-MORE setting if you wish.

If you want to create a signature sound for your broadcast or netcast that is far out of the ordinary, if your taste differs from the people who programmed the LESS-MORE tables, or if you are using OPTIMOD-PCn in mastering or production applications, you will find the advanced processing controls useful. You can customize or modify any subjective setup control setting to create a sound exactly to your taste. You can then save the settings in a User Preset and recall it later. This sort of customization is usually unnecessary for sound-for-picture but can be very useful for radio, production, and mastering applications.

Compressor attack time, release time, and threshold controls are available. These controls can be dangerous in inexperienced hands, leading you to create presets that sound great on some program material but overdrive the peak limiter on other material, causing objectionable pumping or distortion. We therefore recommend that you create custom presets at the Advanced Modify level only if you are experienced with audio processing sound design and if you are willing to take the time to double-check your work on many different types of program material.

In production and mastering applications, you will usually be working with one piece of program material at a time. Here, you can use all of Advanced Modify's power to get the sound you want without being concerned about how your settings will sound with other material.

The PC Remote software organizes its controls in tabbed screens. The EQUALIZATION, STEREO ENHANCER, and LESS-MORE labs access the Basic Modify controls. The remaining tabs show the Advanced Modify controls, logically organized by functionality.

> **Important Note:** Once you have edited a preset's dynamics parameters in Advanced Modify, LESS-MORE control is no longer available in Basic Modify. As noted above, we strongly recommend using the LESS-MORE control to achieve a sound as close as possible to your desired sound before you make further modifications at the Advanced Modify level.

Gain Reduction Metering

Because it uses floating point processing, OPTIMOD-PCn has essentially unlimited amounts of available gain reduction, unlike Orban's DSP-based processors. However, the meter should never exceed 25 dB gain reduction if OPTIMOD-PCn has been set up for a sane amount of gain reduction under ordinary program conditions. If any AGC or compressor gain reduction meter reads full-scale, this is usually a sign that you are using too much gain reduction, which can cause unpleasant compression artifacts.

The peak limiter gain reduction meters have 12 dB of range. If you are using the MX limiter and the input program material is not excessively peak-limited, it is common for these meters to go to full-scale briefly without audibly objectionable consequences. However, you must assess this by ear for the program material you are processing.

Fundamental Requirements: High-Quality Source Material and Accurate Monitoring

A major potential cause of distortion is excess peak limiting. Another cause is poor-quality source material, including the effects of your playback machines, electronics, and studio-to-transmitter link (if any). If the source material is distorted even slightly, that distortion can be exaggerated by OPTIMOD-PCn—particularly if a large amount of gain reduction is used. Very clean audio can be processed harder without producing objectionable distortion.

A high-quality monitor system is essential. To modify your sound effectively, you must be able to hear the results of your adjustments. In too many facilities, the best monitor is significantly inferior to the sound systems found in many listeners' homes!

Factory Programming Presets

Factory Programming Presets are our "factory recommended settings" for various program formats or types. The Factory Programming Presets are starting points to help you get on the air quickly without having to understand anything about adjusting OPTIMOD-PCn's sound.

You can easily edit any of these presets with the LESS-MORE control to optimize the trade-off between loudness and distortion according to the needs of your format, although this is often unnecessary. It is OK to use unmodified factory presets on the air. These represent the best efforts of some very experienced transmission processing sound designers. We are sometimes asked about unpublished "programming secrets" for Optimods. In fact, there are no "secrets" that we withhold from users. This manual reveals our "secrets" and the presets embody all of our craft as processing experts. The presets are editable because other sound designers may have different preferences from ours, not because the presets are somehow mediocre or improvable by those with special, arcane knowledge that we withhold from most of our customers.

Start with one of these presets. Spend some time listening critically to your sound. Listen to a wide range of program material typical of your format and listen on several types of audio systems (not just on your studio monitors). Then, if you wish, customize your sound using the information in the Protection Limiter, Two-Band and Five-Band sections that follow.

Each factory preset has full LESS-MORE capability. The table shows the presets, including the source presets from which they were taken and the nominal LESS-MORE setting of each preset. Some of the Five-Band presets appear several times under different names because we felt that these presets were appropriate for more than one format; these can be identified by a shared source preset name.

> **Important!** If you are dissatisfied with the sound available from the factory presets, please understand that each named preset is actually 19 presets that can be accessed via the LESS-MORE control. Try using this control to trade off the amount of dynamic range reduction against processing artifacts and side effects. Once you have used LESS-MORE, save your edited preset as a User Preset.

Do not be afraid to choose a preset other than the one named for your programming if you believe this other preset has a more appropriate sound. Also, if you want to fine-tune the frequency balance of the programming, feel free to use Basic Modify and make small changes to the Bass, Mid EQ, and HF EQ controls. OPTIMOD-PCn lets you make changes in EQ (and stereo enhancement) without losing the ability to use Less-More settings.

> Of course, Less-More is still available for the unedited preset if you want to go back to it. Other than by purposeful abuse within Windows (the preset files have the read-only attribute and this should not be changed), there is no way you can erase or otherwise damage the Factory Presets. So, feel free to experiment.

Protection, AGC and Mastering Presets

PROTECTION, AGC AND MASTERING PRESETS			
Preset Names	**Source Preset**	**Less-More**	**Loudness**
AGC+ LIMIT FLAT	AGC+ LIMIT FLAT	5.0	−15 LUFS
AGC+ LIMIT PRE-E	AGC+ LIMIT 50us	5.0	−15 LUFS
ROCK MASTERING MXH	ROCK MASTERING MXH	5.0	−6 LUFS
ROCK MASTERING MX	ROCK MASTERING MX	5.0	−6 LUFS
PEAK LIMIT MX	PEAK LIMIT MX	1.0	−16 LUFS
SOFT KNEE	SOFT KNEE	5.0	−12 LUFS
SOFT KNEE MX	SOFT KNEE MX	5.0	−12 LUFS

Table 3-1: Protection, AGC and Mastering Presets

AGC+LIMIT [FLAT, PRE-E]: These presets allow OPTIMOD-PCn to serve as a studio AGC, substituting for the AGC in an Optimod at a radio or television transmitter and providing protection limiting for the STL that links the output of OPTIMOD-PCn to the input of the Optimod at the transmitter. See *Studio AGC* on page 3-9.

If PROC PRE-EMPHASIS (in I/O > STEREO OUTPUT) is FLAT, use the AGC+LIMIT FLAT preset; otherwise, use AGC+LIMIT PRE-E. See step (4.C) on page 2-18. These presets are identical, except that AGC+LIMIT PRE uses the MX limiter and AGC+LIMIT FLAT does not.

(The MX limiter is much less vulnerable to audible gain pumping caused by large amounts of preemphasized high frequency energy.)

It is common to adjust the AGC attack times, release times, etc. to your preference and then save the result as a user preset.

PEAK LIMITER MX: The PEAK LIMITER MX preset turns off all processing other than the peak limiter to implement a fast true-peak protection limiter with very high performance.

In the LIMITER tab, you may choose either SOFT or HARD MX overshoot limiter modes. HARD can be useful if you need to produce maximum loudness. The highest usable loudness available in HARD mode is approximately −4 LUFS, but we recommend using −6 LUFS or lower to achieve best quality. Attempting to go above −6 LUFS quickly sacrifices bass punch.

The subjective differences between SOFT and HARD are subtle. SOFT sounds cleaner on most material, but can produce some audible gain pumping when driven very hard. HARD is closer to a "clipper sound," so it is not quite as clean sounding but does not produce gain pumping. It may exhibit less IM distortion on material with very heavy bass, particularly with dominant frequencies below 50 Hz.

The loudness meters continue to work, and continue to be calibrated such that 0 is equal to the setting of the TARGET LOUDNESS control. However, they are located before the peak limiters, so if you are driving the peak limiters hard so that their limiting action alone changes loudness, you should monitor the output of OPTIMOD-PCn with an external loudness meter like the free Orban loudness meter, which can be bridged across the OPTIMOD-PCn output. (Get the meter at www.orban.com/meter.)

Warning: If you use HARD mode with the OPTIMIX DEFEAT SWITCH (in I/O > UTILITIES) set to ACTIVE, you may overload your computer's CPU and cause audio stuttering. The stuttering will normally stop if you change the control setting that caused it. Worst-case, clearing this may require stopping and restarting the Service, which you can do by disconnecting 1600PC from any Processor and then entering the TOOLS > SERVICE SETTINGS dialog box and clicking RESTART. Only the fastest computers can run HARD mode combined with Optimix, but many computers can use HARD MODE with Optimix defeated. You can also get more CPU headroom by setting the CBS LOUDNESS CONTROLLER switch (in I/O > UTILITIES) to OUT.

SOFT KNEE: *These presets (in MX and non-MX versions) are "zeroed-out" starting points for mastering applications*, particularly where soft knee compression is desired. *These presets must be manually tweaked to complement the program material being processed*. See *Production and Mastering* starting on page 3-86.

These presets are phase-linear. They set all equalization flat and turn off the AGC. The multiband compressor (two-band or five-band) is set to supply a very soft-knee compression characteristic with approximately 5-10 dB of gain reduction. The ratio for a given compressor starts out at 1:1 and ends up at ∞:1 when the input level is 20 dB above threshold.

Mastering engineers will certainly want to adjust the compression thresholds and band coupling to complement the program material. The ratio and knee controls are adjustable separately for each band's compressor. For example, one might want to use a low ratio and soft knee in bands 1-4 while using a higher ratio and/or harder knee in band 5 (for de-essing).

The 1600's powerful equalization section is, of course, also available. Additionally, these presets set the look-ahead limiter drive control conservatively, which ensures highest quality. However, the 1600's look-ahead limiter can be driven quite hard without objectionable side effects, so the 1600 can create competitively loud masters. For "loudness war" masters, start with the MX version.

ROCK MASTERING [MX, MXH]: These presets demonstrate how OPTIMOD-PCn can be used as a mastering processor to make clean, bright, detailed, punchy, and very loud masters. They are based on the SOFT KNEE MX preset, but with substantial customization. They work best when fed mixes that have not previously had dynamics processing (such as wideband compression) applied to the entire mix.

These presets are a useful starting point, but we expect that any mastering engineer would tweak them for each piece of program material being mastered. For an in-depth discussion, see *Production and Mastering* starting on page 3-86.

Rock Mastering MXH uses HARD MX LIMIT OVERSHOOT MODE. Note the **Warning** on page 2-35; this preset can overload the CPU of many computers.

Radio-Style Presets

RADIO-STYLE PRESETS			
Preset Names	**Source Preset**	**Less-More**	**Loudness**
Adult Contemp MX MED	GREGG HD MX Med XLF	9.0	−7 LUFS
Adult Contemp MX OPEN	GREGG HD MX Open XLF	9.0	−10 LUFS
Alternative MX	GREGG HD MX Med XLF	9.0	−7 LUFS
Ambient MX LIGHT	GREGG HD MX Light	7.0	−11 LUFS
Ambient MX OPEN	GREGG HD MX Open XLF	9.0	−10 LUFS
Chill MX LIGHT	GREGG HD MX Light	7.0	−11 LUFS
Chill MX OPEN	GREGG HD MX Open XLF	9.0	−10 LUFS
CHR MX HEAVY	GREGG HD MX Xcess XLF	9.0	−6 LUFS
CHR MX MED	GREGG HD MX Med XLF	9.0	−7 LUFS
CLASSICAL-5 BAND MX	CLASSICAL-5 BAND MX	7.0	−14 LUFS
CLASSICAL-5 BAND	CLASSICAL-5 BAND	7.0	−14 LUFS
CLASSICAL-5 BAND Parallel	CLASSICAL-5 BAND Parallel	5.0	−14 LUFS
CLASSICAL-5B+AGC MX	CLASSICAL-5B+AGC MX	5.0	−14 LUFS
CLASSICAL-5B+AGC	CLASSICAL-5B+AGC	5.0	−14 LUFS
CLASSICAL-5B+AGC Parallel	CLASSICAL-5B+AGC Parallel	5.0	−14 LUFS
Classic Hits MX	GREGG HD MX Heavy XLF	10.0	−7 LUFS
Classic Rock MX	GREGG HD MX Med XLF	9.0	−7 LUFS
Country MX	GREGG HD MX Med XLF	9.0	−7 LUFS

RADIO-STYLE PRESETS			
Preset Names	**Source Preset**	**Less-More**	**Loudness**
Dance MX	GREGG HD MX XXLF	9.0	−8 LUFS
Easy Listen MX LIGHT	GREGG HD MX Light	7.0	−11 LUFS
Easy Listen MX OPEN	GREGG HD MX Open XLF	9.0	-10 LUFS
GOLD MX	GOLD MX	9.5	−8 LUFS
GOLD	GOLD	5.0	−10 LUFS
Gold MX HEAVY	GREGG HD MX Heavy XLF	10.0	−7 LUFS
GREGG HD MX Med Dry	GREGG HD MX Med Dry	9.0	−9 LUFS
GREGG HD MX Med Dry XLF	GREGG HD MX Med Dry XLF	9.0	−9 LUFS
GREGG HD MX Heavy	GREGG HD MX Heavy	7.0	−7 LUFS
GREGG HD MX Heavy XLF	GREGG HD MXH Heavy XLF	10.0	−5 LUFS
GREGG HD MXH Heavy XLF	GREGG HD MX Heavy XLF	10.0	−7 LUFS
GREGG HD MX Light	GREGG HD MX Light	7.0	−11 LUFS
GREGG HD MX Light XLF	GREGG HD MX Light XLF	7.0	−11 LUFS
GREGG HD MX Med	GREGG HD MX Med	9.0	−7 LUFS
GREGG HD MX Med XLF	GREGG HD MX Med XLF	9.0	−7 LUFS
GREGG HD MX Open	GREGG HD MX Open	9.0	-10 LUFS
GREGG HD MX Open XLF	GREGG HD MX Open XLF	9.0	-10 LUFS
GREGG HD MX Xcess	GREGG HD MX Xcess	9.0	−6 LUFS
GREGG HD MX Xcess XLF	GREGG HD MX Xcess XLF	9.0	−6 LUFS
GREGG HD MXH Xcess XXLF	GREGG HD MXH Xcess XXLF	10.0	–7 LUFS
GREGG HD MX XXLF	GREGG HD MX XXLF	9.0	−8 LUFS
GREGG HD MX XXXLF	GREGG HD MX XXXLF	10.0	–8 LUFS
GREGG HD Med Dry XLF	GREGG HD Med Dry XLF	9.0	−11 LUFS
GREGG HD Med Dry	GREGG HD Med Dry	9.0	−11 LUFS
GREGG HD Heavy	GREGG HD Heavy	7.0	−9 LUFS
GREGG HD Heavy XLF	GREGG HD Heavy XLF	10.0	−9 LUFS
GREGG HD Light	GREGG HD Light	7.0	−11 LUFS
GREGG HD Light XLF	GREGG HD Light XLF	7.0	−11 LUFS
GREGG HD Med	GREGG HD Med	9.5	−9 LUFS
GREGG HD Med XLF	GREGG HD Med XLF	9.0	−9 LUFS
GREGG HD Open	GREGG HD Open	9.0	−10 LUFS
GREGG HD Open XLF	GREGG HD Open XLF	9.0	−10 LUFS
GREGG HD Xcess	GREGG HD Xcess	9.0	−8 LUFS
GREGG HD Xcess XLF	GREGG HD Xces XLF s	9.0	−8 LUFS
GREGG HD XXLF	GREGG HD XXLF	9.0	-10 LUFS
Hip-Hop MX	GREGG HD MX XXLF	9.0	−8 LUFS
Indie MX	GREGG HD MX Med XLF	9.0	−7 LUFS
Jazz MX	POP MX OPEN FLAT	9.0	−10 LUFS
Lounge MX LIGHT	GREGG HD MX Light	9.0	−10 LUFS
Lounge MX OPEN	GREGG HD MX Open XLF	9.0	−10 LUFS
Modern Rock MX	GREGG HD MX Med XLF	9.0	−7 LUFS
NEWS-TALK MX	NEWS-TALK MX	7.0	-10 LUFS
NEWS-TALK	NEWS-TALK	7.0	−10 LUFS
Oldies MX	GREGG HD MX Heavy XLF	10.0	−7 LUFS
POP MX XLF	POP MX XLF	9.0	−8 LUFS
POP MX FLAT EQ	POP MX FLAT EQ	9.0	−8 LUFS
POP MX LIGHT FLAT	POP MX LIGHT FLAT	7.0	−11 LUFS
POP MX OPEN FLAT	POP MX OPEN FLAT	9.0	−10 LUFS
POP XLF	POP XLF	9.0	−10 LUFS
POP FLAT EQ	POP FLAT EQ	9.0	−10 LUFS
POP LIGHT FLAT	POP LIGHT FLAT	7.0	−11 LUFS

RADIO-STYLE PRESETS			
Preset Names	**Source Preset**	**Less-More**	**Loudness**
POP OPEN FLAT	POP OPEN FLAT	9.0	−10 LUFS
R and B MX	GREGG HD MX XXLF	9.0	−8 LUFS
Rap MX	GREGG HD MX XXLF	9.0	−8 LUFS
Reggae MX	GREGG HD MX XXLF	9.0	−8 LUFS
Rock MX	GREGG HD MX Med XLF	9.0	−7 LUFS
Smooth Jazz MX LIGHT	GREGG HD MX Light	7.0	−11 LUFS
Smooth Jazz MX OPEN	GREGG HD MX Open XLF	9.0	-10 LUFS
Soul MX	GREGG HD MX Heavy XLF	10.0	−7 LUFS
Soul MX XXLF	GREGG HD MX XXLF	9.0	−8 LUFS
SPORTS MX	SPORTS MX	7.0	−10 LUFS
SPORTS	SPORTS	7.0	− 10 LUFS
Standards MX LIGHT	POP MX LIGHT FLAT	7.0	−11 LUFS
Standards MX OPEN	GREGG HD MX Open XLF	9.0	−10 LUFS
Urban	GREGG HD MX XXLF	9.0	−8 LUFS

Table 3-2: Radio-Style Presets

- If a preset's name contains "XLF," the subharmonic synthesizer is active.
- If a preset's name contains "MX," the MX peak limiter is active in SOFT overshoot limiting mode. If a preset's name contains "MXH," the MX peak limiter is active in HARD overshoot limiting mode.
- Most presets with names that are specific to a programming format (like "Modern Rock") duplicate other presets, called the "Source Presets" in Table 3-2. The format-specific duplicates were created solely for your convenience and are *only suggestions.* You may prefer the sound of a preset whose name is not the same as your format; this is perfectly OK.

  All of the presets with format-specific names use the MX limiter. To create a non-MX format-specific preset, use the non-MX version of the format-specific preset's "Source Preset."

CLASSICAL: As their names imply, the CLASSICAL 5-BAND and CLASSICAL 5-BAND+AGC presets (in MX and non-MX versions) are optimized for classical music, gracefully handling recordings with very wide dynamic range and sudden shifts in dynamics. They use heavy inter-band coupling to prevent large amounts of automatic re-equalization, which could otherwise cause unnatural stridency and brightness in strings and horns and which could pump up very low frequency rumble in live recording venues.

The Five-Band preset defeats the AGC, using only the five-band compressor for gain reduction. It also defeats phase rotation and uses the phase-linear five-band crossover to ensure the most transparent Five-Band sound available.

CLASSICAL-5B+AGC uses the AGC set for 2:1 compression ratio. Because of the AGC, it affects more of the total dynamic range of the recording than does the CLASSICAL-5 BAND preset. However, the AGC provides extremely smooth and unobtrusive compression because of the gentle ratio and window gating. This preset uses the Five-Band compressor very lightly with a fast release time as a peak limiter. The AGC does almost all of the compression.

CLASSICAL-5B defeats the AGC and exploits OPTIMOD-PCn's soft-knee five-band compression with an initial compression ratio of 1:1. Quiet material is gently compressed with a very low compression ratio. The compression ratio increases as the source material gets louder (see Figure 3-16 on page 3-71). Very quiet material is typically amplified by 10 dB. This level-dependent compression ratio provides very smooth, subtle compression.

CLASSICAL-5B+AGC PARALLEL and CLASSICAL-5 BAND PARALLEL use the compressor's parallel mode to bring up low-level passages while preserving more and more of the music's dynamics as it gets louder (see Parallel Compression on page 3-21). These presets can be very subtle and musical while still allowing the quiet parts of the music to be heard clearly in noisy environments.

For these presets, LESS-MORE sets the amount of amplification of quiet parts of the program. While neither Parallel preset uses the MX limiter (because the low target loudness typically used in classical operations usually causes little or no peak limiting), you can activate the MX limiter without losing LESS-MORE functionality.

Because these presets pass the dynamics of loud parts of the music through essentially unmodified, you must use care not to overdrive the peak limiter. These presets are safest to use with low target loudness (–16 LUFS and below), where overdriving the peak limiters is less likely. Because of this level sensitivity, we expect that most users will want to customize the Parallel presets by adjusting the COMPRESSION THRESH OFFSET control and possibly the TARGET LOUDNESS and/or MB FINAL LIMIT DRIVE controls.

CLASSICAL-5B+AGC PARALLEL has the AGC turned on to prevent the peak limiters from being grossly overdriven if the input is unusually loud. However, because of the AGC's slow attack time and 2:1 compression ratio, loud transients can set through

CLASSICAL-5 BAND PARALLEL has the AGC turned off, so operators must do appropriate gain riding to prevent annoying loudness inconsistencies between different recordings. In facilities where the program is played out via an automation system, it may be sufficient to statically normalize the gain of individual elements so that the BS.1770 Integrated Loudness of the entire element meets a target value, typically –16 to –20 LUFS. With classical music formats, the target loudness of speech material should be 3 to 6 LU below the target loudness of music.

Because the CLASSICAL presets preserve a significant amount of the dynamic range present in the source material (including speech), it is wise to use a separate microphone processor to ensure appropriate voice/music balance.

GOLD [MX]: GOLD and GOLD MX are loud and "hi-fi"-sounding while still respecting the limitations and basic flavor of the recordings from the era of the 1950s through 1970s. It is the only OPTIMOD-PCn preset that uses a considerable amount of fixed equalization to achieve a bright, punchy, "present" sound without over-exaggerating high frequencies. The highs in recordings of this era are often noisy, distorted, or have other technical problems that make them unpleasant sounding when the processor over-equalizes them in an attempt to emulate the high frequency balance of recently recorded material.

Because they provide an example of using fixed parametric equalization, these presets can be useful as a guide for further customization. (Most of the OPTIMOD-PCn radio-style presets do not use fixed equalization other than shelving bass boost.)

The **GREGG** family of presets was designed by Greg Ogonowski. Most use a 200 Hz band1/band2 crossover frequency to achieve a bass sound similar to the classic five-band Gregg Labs FM processors. Dynamically, most of these presets produce a considerable increase in bass energy below 100 Hz and a decrease of bass energy centered at 160 Hz. This bass sound works particularly well with speakers having good bass response.

The GREGG HD MX XXX presets fully exploit OPTIMOD-PCn's MX peak limiter technology to be loud without the usual peak limiter artifacts. There are also several non-MX presets, which can be used when the target loudness is lower and it is important to maximize the number of instances of OPTIMOD-PCn that can run on a given computer; these presets cut CPU usage approximately in half compared to the MX presets.

Most of the GREGG presets are designed to achieve extremely punchy bass, even with program material that lacks bass. GREGG presets use the subharmonic synthesizer in conjunction with bass equalization to achieve this. The processing is tuned dynamically (using the bass limiter and Band 1 of the multiband compressor) to prevent bass from becoming excessive if the program material already has sufficient bass.

As is always true with presets, the only way to judge them and decide if they are appropriate for your goals is to audition them with program material typical of your format.

GREGG HD and GREGG OPEN HD are good general-purpose presets for popular music programming if you wish to save CPU cycles by not using the MX limiter. The OPEN version uses a slower multiband release time and increases density less.

Sonically, GREGG HD and GREGG OPEN HD are not competitive with the GREGG HD MX xxx presets unless the target loudness is −16 LUFS or lower (above that, the MX presets will sound brighter, more detailed, and punchier). However, GREGG HD and GREGG OPEN HD will not produce overtly objectionable artifacts (such as audible gain pumping) unless the target loudness is −8 LUFS or higher.

GREGG HD MX HEAVY, MEDIUM, and LIGHT are good general-purpose presets for popular music programming if you want an excellent trade-off between loudness and artifacts. As you go from LIGHT to HEAVY, OPTIMOD-PCn will produce more and more density and source-to-source consistency. GREGG HD MX MEDIUM is the factory default preset in a new OPTIMOD-PCn installation.

GREGG HD MX OPEN uses a SLOW multiband release to achieve a combination of open, relaxed sound and good source-to-source consistency.

GREGG HD MX XCESS is an "up-against-wall" preset that is nevertheless useful if you want a very consistent, high-energy presentation—despite the implication of "excess," this is not a "joke preset." (There no useless, "joke presets" in OPTIMOD-PCn!)

GREGG HD MX XXLF goes for extreme bass punch and impact for formats like Urban and Dance, and was originally created for European customers seeking an "over the top" bass texture.

GREGG HD MXH Heavy XLF uses HARD MX clipping to achieve extremely high loudness. With turned-down down the LIMITER ATTACK controls, It is purposely designed to exhibit some compressor pumping, giving it a more "processed" sound than the other GREGG presets.

GREGG HD MX XXXLF is similar to GREGG HD MX XXLF. This preset generates a lot of synthetic bass energy and is targeted towards "Gold" and "Oldies" formats where the original program material lacks bass impact.

GREGG HD MXH Xcess XXLF was derived from GREGG HD MX Xcess, but uses HARD MX limiting and, like GREGG HD MX XXXLF, generates generous amounts of synthetic bass and is optimized for "Gold" and "Oldies" formats.

NEWS-TALK: This preset (in both MX and non-MX versions) is quite different from the others. It is based on the fast multiband release time setting so it can quickly perform automatic equalization of substandard program material, including telephone. It is useful for creating a uniform, intelligible sound from widely varying source material, particularly source material that is "hot from the field" with uncontrolled quality.

The MX limiter has little value for speech programming, as the non-MX look-ahead limiter is very clean on speech. NEWS-TALK MX exists mostly for the convenience of users having mixed formats, streaming music for part of the day and news or talk at other times. It allows you to use an MX preset for the music programming and switch between the music and news-talk presets without audible glitches caused by a sudden change in input/output delay when activating the MX limiter.

The MX limiter adds about 240 ms of delay when active.

POP MX: The POP MX family of presets offer a different bass texture compared to the GREGG presets. In general, they do not push low bass as hard, and may therefore be preferred to the GREGG presets if most of the audience is listening on player devices with small loudspeakers having no significant response below 100 Hz or so.

POP+SUBHARM MX is appropriate for programming that mixes recordings from different eras. It uses the subharmonic synthesizer to add low bass to material that lacks it. As such, it is the POP preset most appropriate for player devices with good low-frequency response.

POP MX FLAT EQ, POP MX LIGHT FLAT, and POP MX OPEN FLAT use no bass boost or subharmonic synthesis, although they still exploit the ability of the multiband compressor to automatically re-equalize material that lacks bass. They use a combination of the multiband compressor and HF Enhancer to re-equalize dull-sounding material automatically. The LIGHT version uses SLOW multiband release time to minimize density build-up caused by the multiband compressor. OPEN is very similar to LIGHT, except that OPEN uses the multiband compressor's LINEAR-PHASE crossover mode, whereas LIGHT uses ALLPASS

These presets may be preferred if the musical style of the program material calls for modest amounts of bass. Examples include traditional country and smooth jazz.

SPORTS: Similar to NEWS-TALK except the AGC Release (AGC Release Time) is slower and the Gate Thresh (Gate Threshold) is higher. This recognizes that most sports programming has very low signal-to-noise ratio due to crowd noise and other on-field sounds, so the preset does not pump this up as the NEWS-TALK preset would tend to do.

Sound for Picture Presets

Note on A/V Sync for version 0.9 software: As of this writing, OPTIMOD-PCn v0.9's input/output delay varies from about 450 to 1000 ms, depending on which features are activated. Each time the OPTIMOD-PCn Service is stopped and restarted, the delay may vary by as much as 25 ms due to use of the Windows MME sound model for the input device, and this will make it difficult to achieve consistent A/V sync. In future software releases, we expect to reduce the delay somewhat and to make it more repeatable by making the Windows WASAPI model available at the input.

The sound-for-picture presets have GLOBAL TARGET LOUDNESS. They use non-MX limiter mode because little or no peak limiting is required for typical target loudness used in video applications, so it is pointless to waste CPU cycles on MX limiting.

Most of these presets have the CBS Loudness Controller turned on. It is set so that

SOUND-FOR-PICTURE PRESETS			
Preset Names	**Source Preset**	**Less-More**	**Loudness**
TV 5B DRAMA COUPLED	TV 5B DRAMA COUPLED	5.0	GLOBAL
TV 5B DRAMA	TV 5B DRAMA	5.0	GLOBAL
TV 5B GEN PUR W NR	TV 5B GEN PUR W NR	5.0	GLOBAL
TV 5B GEN PURP NOLC	TV 5B GEN PURP NOLC	5.0	GLOBAL
TV 5B GEN PURPOSE	TV 5B GEN PURPOSE	5.0	GLOBAL
TV 5B NEWS	TV 5B NEWS	5.0	GLOBAL
TV 5B OPTICAL FILM	TV 5B OPTICAL FILM	5.0	GLOBAL
TV 5B SPORTS	TV 5B SPORTS	5.0	GLOBAL
TV AGC+LC	TV AGC+LC	5.0	GLOBAL
TV AGC+LC+DS	TV AGC+LC+DS	5.0	GLOBAL

Table 3-3: Sound-For-Picture Presets

loudness is constrained to a level consistent with the active TARGET LOUDNESS value (see *Loudness Control* on page 3-24). The TARGET LOUDNESS value is the same as the "target loudness" as measured by an ITU-R BS.1770-2 (or higher) Integrated loudness meter (with a 10-second rolling windows of integration) in units of LkFS (ATSC Rec. A/85) and LUFS (EBU Rec. R128). To comply with these standards, TARGET LOUDNESS is customarily set to −24 in the United States and −23 in the EU.

To constrain the reading of a BS.1770 loudness meter (included with OPTIMOD-PCn) to a fixed threshold, almost all "TV" presets have the BS.1770 THRESHOLD control set to 0 LK. However, if your organization does not have a strict policy about processing for the BS.1770 meter, we recommend that you edit your preferred preset by setting this control anywhere from +2 to OFF and then saving the result as a User Preset. For a more detailed discussion, see *BS.1770 Safety Limiter* on page 3-26.

If you wish to use a given preset with the CBS loudness controller turned off, set the LOUDNESS THRESHOLD control (in the MULTIBAND tab) to OFF and save the result as a User Preset.

TV 5B-GEN PUR W/NR (TV Five-Band General Purpose with Noise Reduction): provides effective dynamic range control and "automatic re-equalization" of most dramatic material. It uses the Loudness Controller to control loudness tightly. It applies single-ended noise reduction to the material, which will reduce unwanted noise like hiss, hum, or stage rumble. However, it will also reduce ambience. If the program material is carefully produced (as are most contemporary feature-film soundtracks), you may wish to use TV 5B-GEN PURPOSE (which does not apply noise reduction), or, if the material is so well produced that it would not benefit from "automatic re-equalization," use TV 2B-GEN PURPOSE.

TV 5B-GEN PURPOSE (TV Five-Band General Purpose without Noise Reduction): is identical to TV 5B-GEN PUR W/NR except that the single-ended dynamic noise reduction system is off.

TV 5B GEN PURP −LC (TV Five-Band General Purpose without Loudness Controller): This preset is the same as TV 5B-GEN PURPOSE except the Loudness Controller is turned off and the MB DRIVE control is backed off by 3 dB to achieve approximately the same loudness as TV 5B-GEN PURPOSE. Because the Loudness Controller is inactive, this preset has more dynamic punch than TV 5B-GEN PURPOSE. The five-band processing makes the audio spectrum more consistent than does TV 2B-GEN PUR −LC, so TV 5B-GEN PURP −LC controls loudness better than TV 2B-GEN PURP −LC even though the Loudness Controller is inactive.

TV 5B-DRAMA (TV Five-Band Drama): uses the 1600's soft knee compression and AGC RATIO control to regulate loudness while still preserving some of the dynamic range of the original mix. In addition, the five-band compressor automatically re-equalizes material that may otherwise sound spectrally unbalanced. The center channel compressor effectively de-esses dialog without punching holes in the remaining channels. Coupling between the center channel and remaining channels allows the center channel's level to be boosted automatically by as much as 3 dB with respect to the other channels if this is needed to help intelligibility.

We prefer this preset to TV 2B-DRAMA because of its effective, unobtrusive de-essing and because it is more resistant to spectral gain intermodulation, which is program material in one frequency range's audibly pumping material in a different range.

Because it preserves some dynamic range, it is important that the input level to the 1600 not be too far awry. Usually, network feeds will meet this requirement but local playout of older material that has not been checked for loudness by a long-term Loudness Level meter like ITU-R BS.1770 may not work well. Use one of the general-purpose presets for this kind of material.

This preset fully exploits the Loudness Controller, which usually shows slight gain reduction with dialog at normal levels.

TV 5B-DRAMA COUPLD (TV Five-Band Drama Coupled): is similar to TV 5B DRAMA but uses more interband band coupling in the five-band compressor and a slower multiband compressor release time so it performs less automatic re-equalization of the program material. This is an alternative to the two-band compression available on older Optimods.

TV 5B-NEWS (TV Five-Band News): rides gain more quickly than the general-purpose presets. Its AGC release time is faster, so it will bring up low-level material more quickly. It is designed for live news programs where input levels may be quite unpredictable. It also automatically re-equalizes substandard audio (which is quite common in live news broadcasts). The dynamic single-ended noise reduction is turned on.

TV 5B-SPORTS (TV Five-Band Sports): is similar to TV 5B-NEWS, except the AGC release time is slower to resist pumping up crowd noise.

TV 5B OPTICAL FILM (TV Five-Band Optical Film): is designed to make the best of the low-quality audio provided with optical film sound tracks (particularly 16mm). The gate threshold is quite high to avoid pumping up hiss, thumps, and other optical artifacts. The threshold of the single-ended dynamic noise reduction system is also high so that this system can reduce artifacts as far as possible. Release times are slow, because we assumed that material encoded on optical film has already been carefully level-controlled to accommodate the very limited dynamic range of the medium, so little gain riding is therefore required from the 1600.

Editing Presets

OPTIMOD-PCn is very flexible, enabling you to fine-tune your sound to your requirements. Each preset can be edited with the LESS-MORE control. In all presets, LESS-MORE attempts to keep the target loudness constant and equal to the active TARGET LOUDNESS value, which is either local to the preset, or if the preset's TARGET LOUDNESS control is set to GLOBAL, the TARGET LOUDNESS specified in I/O SETUP > GLOBAL.

If you want to create your own User Presets, the following detailed discussion of the processing structures is important to understand. If you only use Factory Presets or if you only modify them with LESS-MORE, you may still find the material interesting but you do not need to understand it to get excellent sound from OPTIMOD-PCn. We have carefully designed OPTIMOD-PCn's factory presets and most users will not need to go beyond these.

The LESS-MORE control affects the video-oriented presets differently than it does the music presets. When a video-oriented preset is on the air, the LESS-MORE control adjusts the average amount of gain reduction by adjusting the AGC DRIVE control. This also adjusts the *idle gain*—the amount of gain reduction in the AGC section when the structure is gated. (It gates whenever the input level to the structure is below the user-adjustable threshold of gating.)

When a music preset is on the air, the LESS-MORE control sets the amount of overall processing while attempting to keep the target loudness constant. Typically, it does this by adjusting the MULTIBAND DRIVE and MB FINAL LIMIT DRIVE controls in a complementary way: If the MULTIBAND DRIVE IS turned up, the MB FINAL LIMIT DRIVE is turned down. Hence, with music presets OPTIMOD-PCn's loudness meters show how LESS-MORE affects loudness.

> Because the music presets do not use OPTIMOD-PCn's CBS Loudness Controller or BS.1770 Safety Limiter, the loudness meters will show more variation than they do for the video presets. However, we believe that subjectively, they provide excellent consistency of both loudness and texture between program elements.

The controls give you the flexibility to customize your sound. However, as with any audio processing system, proper adjustment of these controls requires proper balancing of the trade-offs explained above. The following provides the information you need to adjust controls to suit your applications and taste.

Optimix™ Upmixer Controls

Optimix Adjustments	
Name	**Range**
Mode	Bypass; Auto-Detect; Upmix ON
Center Width	0 to 1 in 0.1 steps
Front/Rear Balance	0 to 1 in 0.1 steps
Stage Shape	Frontal, Wraparound
Center Delay Correction	ON, OFF

Table 3-4: Optimix Upmixer Adjustments

These controls are only available if the Processor's output has been configured for 5.1 channels. See step 10.B)c) on page 2-9.

Optimix automatically upmixes stereo program material above 80 Hz to 5.0 surround that is fully compatible with 5.1 systems. Because of the wide variability in subwoofer setups, Optimix instead routes energy below 80 Hz from the original stereo mix to the Lf and Rf outputs of the upmixer. If properly set up, a home theater receiver will route this information to a subwoofer if one is in place and the Lf and RF loudspeakers are "small."

Delay Correction (ON, OFF) automatically corrects time skew or phase shift between the left and right input channels. It prevents comb filtering in the derived center channel. This maximizes mono compatibility and maintains the best possible crispness and intelligibility in the center channel, which usually contains dialog. Because the left/right phase corrector is automatically bypassed when the Optimod MODE is AUTO or UPMIX, you must use the Optimix delay corrector if you wish to phase-correct an upmix.

Optimix has two controls that affect the shape of the upmixed surround soundstage.

Center Width (0 to 1 in 0.1 steps) determines whether extracted center-channel information will be routed solely to the center output or if some amount of the center energy is blended into the Left Front and Right Front outputs. While setting this control to "0" maximizes the "discreteness" of the upmix, some operators prefer to mix a certain amount of center energy into Lf and Rf to ensure that dialog is not lost completely if the center channel is not correctly set up at the consumer's receiver.

Front/Rear Balance (0 to 1 in 0.1 steps) is the ratio between the Lf/Rf and Ls/Rs levels above 80 Hz, in absolute (not dB) units. The control "pans" the derived left/right channels (where the center information has been removed) between the Lf/Ls and Rf/Rs output pairs respectively. Technically, it is used to determine the amount of envelopment in the upmix. Setting this control to 1.0, which mixes the derived L/R channels equally front and rear, works well for television-oriented audio. Setting the control to 0 turns off the surround channels.

Stage Shape (WRAP, FRONTAL) determines whether the virtual soundstage is wrapped around the listener (putting the listener "inside the band") or is placed primarily in the front (putting the listener "inside the auditorium"). In FRONTAL mode, the surround speakers contain mainly reflections and other manifestations of the acoustics of the (real or artificial) space in which the stereo recording was placed by its creator. (Optimix never adds artificial reverb.)

About Downmix Compatibility: Optimix reads downmix parameters (Center and Surround downmix coefficients) from the Optimod and automatically chooses gains for the five upmixed channels. If DELAY CORRECTION is OFF, the downmix in WRAP mode is identical to Optimix's stereo input, assuming a Dolby Lo/Ro downmix. Optimix takes into account the settings of its FRONT/REAR BALANCE and CENTER WIDTH controls. When DELAY CORRECTION is ON, the downmix includes the effect of the center channel delay corrector and can make the stereo downmix crisper and more intelligible than the Optimix's stereo inputs. If the receiver is stereo-only, the bass balance in the downmix will be identical to that of the stereo material applied to Optimix's input.

In FRONTAL mode, the downmix can have slightly more reverb or ambience than the original stereo, although there is no artificial reverb added and the result is usually satisfactory.

Stereo Enhancer Controls	
Name	**Range**
Amount	0.0 ... 10.0
In / Out	Out / In
Ratio Lim	70 … 100%
Depth	0 … 10

Table 3-5: Stereo Enhancer Controls

It is the user's responsibility to make sure that the Optimod's downmix level controls agree with the Center Downmix Level and Surround Downmix Level in the AC3 metadata that is being transmitted to consumers. (Dolby specifies the center default value as −3 dB and the surround default value as −1.5 dB.)

Delay Correction and Dolby Surround: Although delay correction is very effective for most program material, it is incompatible with two-channel input material that was previously Dolby Surround-encoded such that the encoder, before encoding, created the phase-shifted surround channels necessary to create an Lt/Rt output, which can be decoded by the consumer using Dolby Pro Logic to L, C, R, and S.

When Delay Correction is on, Dolby Surround material intended for the surrounds will be reproduced in both the surrounds and center. When Delay Correction is off, material encoded for the surrounds will only appear in the surrounds.

Stereo Enhancer Controls

The stereo enhancer emulates the Orban 222 analog stereo enhancer, which increases the energy in the stereo difference signal (L−R) whenever a transient is detected in the stereo sum signal (L+R). It complements Optimix, increasing the sense of envelopment and space while maintaining excellent downmix compatibility.

The stereo enhancer's gating operates under two conditions.

- The two stereo channels are close to identical in magnitude and phase.

  In this case, the enhancer assumes that the program material is actually mono and thus suppresses enhancement to avoid creating undesired channel imbalance.

- The ratio of L−R / L+R of the enhanced signal tries to exceed the threshold set by the L-R / L+R Ratio Limit control.

  In this case, the enhancer prevents further enhancement in order to prevent excess L−R energy, which can sound unnatural and which can increase multipath distortion in FM broadcasting.

Even when OPTIMOD-PCn is used for surround processing, the stereo enhancer is still available. In this case, it is defeated automatically unless Optimix is actively upmixing. This preserves the dimensionality of true surround input material, but helps Optimix provide a more enveloping surround effect.

The STEREO ENHANCER meter indicates the amount of gain (in dB) that applied to the L−R signal.

> It is unwise to use stereo enhancement with low bitrate codecs. At low bitrates, these codecs use various parametric techniques for encoding the spatial attributes of the sound field. Stereo enhancement can unnecessarily stress this encoding process.

The stereo enhancer has the following controls:

Amount sets the maximum spatial enhancement.

Enhancer In / Out bypasses the stereo enhancer. OUT is equivalent to setting the AMOUNT to 0.

L-R / L+R Ratio Limit sets the maximum amount of enhancement to prevent multipath distortion. However, if the original program material exceeds this limit with no enhancement, the enhancer will not reduce it.

Stereo Synthesizer Controls

Stereo Synthesizer Controls	
Name	**Range**
Mono>Stereo Algorithm	Bypass, Auto, Upmix
In / Out	Wide, Narrow
Mono>Stereo Separation	0 … 10

Table 3-6: Stereo Synthesizer Controls

The stereo synthesizer emulates the classic analog Orban 275A automatic stereo synthesizer[1].

This process creates an artificial stereo difference signal (L−R) by passing the mono input through a multistage allpass filter. After matrixing with the original mono input (which is the L+R signal) to produce the synthesized left and right channels, the result is a "complementary comb filter" whose notches are spaced in frequency in an approximately logarithmic manner. Because only the L−R signal is created artificially, it cancels out of a mono mixdown, making the synthesizer's output completely mono-compatible[2].

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[1] The 275A manual, which provides substantial detail, can be downloaded from ftp://ftp.orban.com/275A/.

[2] Robert Orban: "A Rational Technique for Synthesizing Pseudo-Stereo from Monophonic Sources," J. Audio Engineering Society, Volume 18 Issue 2 pp. 157-164; April 1970

The synthesizer can be activated manually via PC Remote or the API (see Section 5 of the manual). Automatic activation is also available. In AUTOMATIC mode, synthesis from audio on the left input channel will occur if silence is detected on the right input channel. A 187 ms look-ahead delay in the audio path compensates for the delay built into the detector to prevent false triggering on extremely brief right channel pauses. Silence gating prevents triggering unless there is activity on the left channel.

The stereo synthesizer has the following controls, located in the UPMIX page of the GUI:

Mono>Stereo Upmix (BYPASS, AUTO, UPMIX) allows you to manually bypass the synthesizer or force it to always upmix from material on the left channel. AUTO forces upmixing when silence is detected on the right input channel, as described above.

Mono>Stereo Algorithm (WIDE, NARROW) sets the number of notches in the complementary comb filters. WIDE produces fewer notches than NARROW. WIDE produces the most dramatic effect on music, while NARROW prevent speech from becoming unnaturally wide-sounding, and is preferable in sound-for-picture applications.

Mono>Stereo Separation sets the gain applied to the allpass filter chain. 0 suppresses synthesis, while 10 causes the magnitudes of the L+R and L−R signals to be identical.

> Assuming equal gains in the L+R and L−R channels, the mathematics of the process require the phase difference of the L and R channels to be 90 degrees at the frequencies where their magnitudes are equal. This can produce"phasiness" that some people dislike. If the output of the synthesizer is applied to OPTIMOD-PCn's phase correction algorithm, this will remove the "phasiness" at higher frequencies at the expense of putting bumps of up to 3 dB in the frequency response of the mono sum (assuming that MONO>STEREO SEPARATION = 10). Because of this compatibility issue, we recommend instead simply turning down the MONO>STEREO SEPARATION control until the result sounds comfortable and convincing. 7 (where the L−R signal is 70% of the L+R signal) is a good compromise value.

AGC Controls

The AGC smoothly rides gain before the multiband compressor. Unlike AGCs in some other Optimods, only a linear-phase crossover is available and the left and right channels are coupled using RMS summation. The AGC MAXDELTAGR control determines the amount of coupling in stereo processors. While we have removed some features relating to left/right band coupling and sum-and-difference operation, we believe that the new AGC design used in the 1600 is substantially smoother and more transparent than the AGCs used in earlier Optimods.

The AGC may be operated in left/right mode or sum-and-difference mode.

AGC ("AGC Off/On") control activates or defeats the AGC.

It is usually used to defeat the AGC when you want to create a preset with minimal processing (such as a CLASSICAL preset). The AGC is also ordinarily defeated if you are using a studio level controller to protect a transmission link before OPTIMOD-PCn. However, in this case it is better to defeat the AGC globally in System Setup [see step (2.C) on page 2-16].

AGC Drive control adjusts signal level going into the slow dual-band AGC, therefore determining the amount of gain reduction in the AGC. This control also adjusts the "idle gain"—the amount of gain reduction in the AGC section when the structure is gated. (It gates whenever the input level to the structure is below the threshold of gating.)

The total amount of gain reduction in the Five-Band structure is the sum of the gain reduction in the AGC and the gain reduction in the five-band compressor. The total system gain reduction determines how much the loudness of quiet passages will be increased (and, therefore, how consistent overall loudness will be). It is determined by the setting of the AGC DRIVE control, by the level at which the console VU meter or PPM is peaked, and by the setting of the MULTIBAND DRIVE (compressor) control.

AGC Release ("AGC Master Release") control provides an adjustable range from 0.5 dB/second (slow) to 20 dB/second (fast). The increase in density caused by setting the AGC RELEASE control to fast settings sounds different from the increase in density caused by setting the Five-band's MULTIBAND RELEASE control to FAST. You can trade the two off to produce different effects.

Unless it is purposely speeded-up (with the AGC RELEASE control), the automatic gain control (AGC) that occurs in the AGC prior to the multiband compressor makes audio levels more consistent without significantly altering texture. Then the multi-

AGC Controls	
Name	**Range**
AGC Off / On	Off / On
AGC Drive	−10 ... 25 dB
AGC Master Release	0.5, 1.0, 1.5, 2 … 20 dB / S
AGC Bass Release	1 … 10 dB/sec
AGC Gate Threshold	Off, −44 ... −15 dB
AGC Bass Coupling	0 … 24 dB, Off
AGC Window Size	−25 … 0 dB
AGC Window Release	0.5 … 20 dB
AGC Ratio	∞:1, 4:1, 3:1, 2:1
AGC Bass Threshold	−12.0 … 2.5 dB
AGC Idle Gain	−10 … +10 dB
AGC Master Attack	0.2 … 6
AGC Bass Attack	1 … 10
AGC Crossover	Allpass, LinearNoDelay,
AGC Matrix	L/R, sum/difference
AGC MaxDelta GR	0.0 … 24.0 dB, off
Master Delta Thresh	−6.0 … +6.0 dB
Bass Delta Thresh	−6.0 … +6.0 dB

Table 3-7: AGC Controls

band compression audibly changes the density of the sound and dynamically re-equalizes it as necessary: booming bass is tightened; weak, thin bass is brought up; highs are always present and consistent in level.

The various combinations of AGC and compression offer great flexibility:

- Light AGC + light compression yields a wide sense of dynamics, with a small amount of automatic re-equalization.
- Moderate AGC + light compression produces an open, natural quality with automatic re-equalization and increased consistency of frequency balance.
- Moderate AGC + moderate compression gives a denser sound, particularly as the release time of the multiband compressor is sped up.
- Moderate AGC + heavy compression (particularly with a FAST multiband release time) results in a “wall of sound” effect, which may cause listener fatigue.

Adjust the AGC (with the AGC DRIVE control) to produce the desired amount of AGC action, and then fine-tune the compression and clipping with the Five-Band structure’s controls.

AGC Gate (“AGC Gate Threshold”) control determines the lowest input level that will be recognized as program by OPTIMOD-PCn; lower levels are considered to be noise or background sounds and cause the AGC or multiband compressor to gate, effectively freezing gain to prevent noise breathing.

In sound for picture, the setting of the gate threshold controls are quite critical if you want the processing to be undetectable to the audience. If this control is set too low, then OPTIMOD-PCn will pump up quiet sounds such as ambience and underscoring to unnaturally high levels.

There are two independent silence-gating circuits in OPTIMOD-PCn. The first affects the **AGC** and the second affects the **multiband compressor** (2-band or 5-band). Each has its own threshold control.

In the five-band structure, the multiband silence gate causes the gain reduction in bands 2 and 3 of the five-band compressor to move quickly to the average gain reduction occurring in those bands when the gate first turns on. This prevents obvious midrange coloration under gated conditions, because bands 2 and 3 have the same gain.

The multiband gate also independently freezes the gain of the two highest frequency bands (forcing the gain of the highest frequency band to be identical to its lower neighbor), and independently sets the gain of the lowest frequency band according to the setting of the DJ BASS boost control (in the Equalization screen). Thus, without introducing obvious coloration, the gating smoothly preserves the average overall frequency response “tilt” of the multiband compressor, broadly maintaining the “automatic equalization” curve it generates for a given piece of program material.

If the MB GATE THR (Gate Threshold) control is turned OFF, the DJ BASS control is disabled.

AGC Bass Coupling control clamps the amount of dynamic bass boost (in units of dB) that the AGC can provide.

The AGC processes audio in a master band for all audio above approximately 200 Hz and a bass band for audio below approximately 200 Hz. The AGC Master and Bass compressor sidechains operate without internal coupling. The gain reduction in the BASS audio path is either the output of the Bass compressor sidechain or the output of the Master band sidechain. The AGC BASS COUPLING control sets the switching threshold. For example, if the AGC BASS COUPLING control is set to 4 dB and the master gain reduction is 10 dB, the bass gain reduction cannot decrease below 6 dB even if the gain reduction signal from the Bass compressor sidechain is lower. However, the audio path bass gain reduction can be larger than the master gain reduction without limit. In the previous example, the bass gain reduction could be 25 dB

The normal setting of the AGC BASS COUPLING control is 0 dB, which allows the AGC bass band to correct excessive bass as necessary but does not permit it to provide a dynamic bass boost.

Window Size determines the size of the floating “slow zone” window in the master band of the AGC. (The Bass band is not windowed.)

The window works by slowing down changes in the AGC gain reduction that are smaller than the WINDOW SIZE. The window has 2:1 asymmetry around the current AGC gain reduction. For example, if the WINDOW SIZE is set to 4 dB, the window extends 4 dB in the release direction and 2 dB in the attack direction.

If the AGC needs to respond to a large change in its input level by making a gain change that is larger than the window, then the AGC’s attack and release controls determine the AGC’s response time. However, if the change in input level is smaller than the window size, the WINDOW RELEASE control determines the attack and release times. This is usually much slower than the normal AGC time constants. This prevents the AGC from building up density in material whose level is already well controlled.

The previous explanation was somewhat simplified. In fact, the window has “soft edges.” Instead of switching abruptly between time constants, the attack and release times morph smoothly between the setting of the WINDOW RELEASE control and the setting of the AGC master release and attack controls.

The normal setting for the WINDOW SIZE is 3 dB.

Window Release (see WINDOW SIZE above.)

AGC Ratio determines the compression ratio of the AGC. The compression ratio is the ratio between the change in input level and the resulting change in output level, both measured in units of dB.

OPTIMOD-PCn compressor can be operated at a compression ratio as low as 2:1. This can add a sense of dynamic range and is mostly useful for subtle fine arts formats like classical and jazz.

AGC Bass Threshold determines the compression threshold of the bass band in the AGC. It can be used to set the target spectral balance of the AGC.

As the AGC B CPL control is moved towards "100%," the AGC BASS THRESHOLD control affects the sound less and less.

The interaction between the AGC BASS THRESHOLD control and the AGC B CPL control is a bit complex, so we recommend leaving the AGC BASS THRESHOLD control at its factory setting unless you have a good reason for readjusting it.

AGC Idle Gain. The "idle gain" is the target gain of the AGC when the silence gate is active. Whenever the silence gate turns on, the gain of the AGC slowly moves towards the idle gain.

The idle gain is primarily determined by the AGC DRIVE setting—a setting of 10 dB will ordinarily produce an idle gain of −10 dB (i.e., 10 dB of gain reduction). However, sometimes you may not want the idle gain to be the same as the AGC DRIVE setting. The AGC IDLE GAIN control allows you to add or subtract gain from the idle gain setting determined by the AGC DRIVE setting.

You might want to do this if you make a custom preset that otherwise causes the gain to increase or decrease unnaturally when the AGC is gated. For example, to make the idle gain track the setting of the AGC DRIVE control, set the AGC IDLE GAIN control to zero. To make the idle gain 2 dB lower than the setting of the AGC DRIVE control, set the AGC IDLE GAIN control to −2.

AGC Bass Attack sets the attack time of the AGC bass compressor (below 200Hz).

AGC Master Attack sets the attack time of the AGC master compressor (above 200Hz).

AGC Bass Release sets the release time of the AGC bass compressor.

AGC Maximum Delta GR (stereo processing only) approximates the maximum gain difference permitted between the two stereo channels of the AGC. Set it to "0" for perfect stereo coupling. In sum/difference mode it constrains the gain difference between the sum and difference channels.

> In surround processing, the RMS sum of all audio channels controls the gain reduction of the AGC, which is identical in all channels.

AGC Matrix (stereo processing only) allows you to operate the AGC in left/right mode or in sum / difference mode. Usually you will operate in left/right mode. However, sum / difference mode can give a type of stereo enhancement that is different from the enhancement modes offered in the built-in stereo enhancer. This will only work if you allow the two channels of the AGC to have different gains. To do this, set the AGC MAXIMUM DELTA GR control greater than zero.

It is unwise to set this control beyond 3 dB. In FM pre-processing applications, multipath distortion could increase because the amount of L–R energy builds up excessively. We prefer using the built-in stereo enhancer because its built-in gating circuits prevent over-enhancement.

Master Delta Threshold (stereo processing only) allows you to set the difference between the compression thresholds of the sum and difference channels. (This control is only useful when you set the AGC MATRIX to SUM / DIF.) By setting the threshold of the difference channel lower than the sum channel, you can have the AGC automatically produce more gain reduction in the difference channel. This will reduce the separation of material with an excessively wide stereo image (like old Beatles records). To make this work, you must set the MAX DELTA GR control away from zero. For example, to limit an excessively wide image while preventing more than 3 dB difference in gain between the sum and difference channels, set the MAX DELTA GR control to 3.0 and the MASTER DELTA THRESHOLD control to some positive number, depending on how much automatic width control you want the Optimod to perform.

Bass Delta Threshold (stereo processing only) works the same as MASTER DELTA THRESHOLD, but applies to the bass band. You will usually set it the same as MASTER DELTA THRESHOLD.

Equalizer Controls

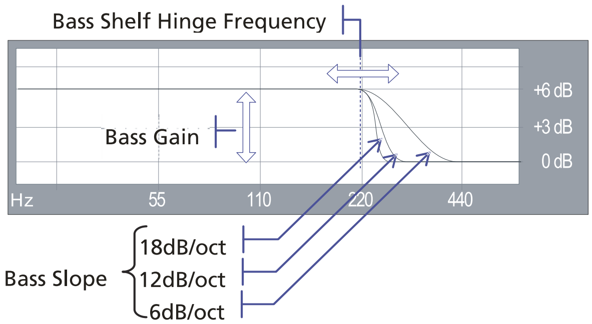
The table summarizes the equalization controls. The equalizer is located between the AGC and five-band compressor sections.

See *Figure 3-6* and *Figure 3-7* on page 3-18 for typical equalizer frequency response curves.

Any equalization that you set will be automatically stored in any User Preset that you create and save. For example, you can use a User Preset to combine an unmodified Factory Programming Preset with your custom equalization. Of course, you can also modify the Factory Preset (with Basic or Advanced Modify) before you create your User Preset.

In general, you should be conservative when equalizing modern, well-recorded program material. This is particularly true with general-purpose video programming.

Except for BASS GAIN, most of the factory presets use less than 3 dB of equalization.



Bass Shelf Controls, the 5-band structure's low bass equalization controls, are designed to add punch and slam to rock and urban music. They provide a parametric shelving equalizer with control over gain, hinge frequency, and slope (in dB/octave).

Bass Shelf Hinge Frequency sets the frequency where shelving starts to take effect.

Bass Gain sets the amount of bass boost (dB) at the top of the shelf.

Bass Slope sets the slope (dB/octave) of the transition between the top and bottom of the shelf.

Because the Five-Band structure often increases the brightness of program material, some bass boost is usually desirable to keep the sound spectrally well balanced. Adjustment of bass equalization must be determined by individual taste and by the requirements of your format. Be sure to listen on a wide variety of consumer systems—it is possible to create severe distortion on poor quality speakers by over-equalizing the bass. Be careful!

The moderate-slope (12 dB/octave) shelving boost achieves a bass boost that is more audible on smaller radios, but which can sound boomier on high-quality systems. The steep-slope (18 dB/octave) shelving boost creates a solid, punchy bass from the better consumer systems with decent bass response. The 6 dB/octave shelving boost is like a conventional tone control and creates the most mid-bass boost, yielding a "warmer" sound. Because it affects the mid-bass frequency range, where the ear is more sensitive than it is to very low bass, the 6 dB/octave slope can create more apparent bass level at the cost of bass "punch."

There are no easy choices here; you must choose the characteristic you want by identifying your target audience and the receivers they are most likely to be using. Regardless of which curve you use, we recommend a +2 to +5 dB boost for most formats. Larger amounts of boost will increase the gain reduction in the lowest band of the multiband compressor, which may have the effect of reducing some frequencies. So be aware the large fixed bass boosts may have a different effect than you expect because of the way that they interact with the multiband compressor. (The GREGG presets use this effect purposely to create a dynamic cut in the mid-bass.)

Low Frequency Parametric Equalizer is a specially designed equalizer whose boost and cut curves closely emulate those of a classic Orban analog parametric equalizer with conventional bell-shaped curves (within ±0.15 dB worst-case). This provides warm, smooth, "analog-sounding" equalization.

LF Freq determines the center frequency of the equalization, in Hertz. Range is 20-500Hz.

LF Gain determines the amount of peak boost or cut (in dB) over a ±10 dB range.

LF Width determines the bandwidth of the equalization, in octaves. The range is 0.8-4.0 octaves. If you are unfamiliar with using a parametric equalizer, 1.5 octaves is a good starting point. These curves are relatively broad because they are designed to provide overall tonal coloration, rather than to notch out small areas of the spectrum.

The LF parametric can be used in the mid-bass region (100-300Hz) to add "warmth" and "mellowness" to the sound when boosting. When cutting, it can remove a "woody" or "boxy" sound.

The equalizer, like the classic Orban model 622B, has constant "Q" curves. This means that the cut curves are narrower than the boost

Equalizer Controls		
Group	**Name**	**Range**
Bass Shelf	Bass Frequency	80, 85, 90, 95, 100, 105, 110, 115, 120, 125, 130, 135, 140, 145, 150, 160, 170, 180, 190, 200, 210, 220, 230, 240, 250, 270, 290, 310, 330, 350, 380, 410, 440, 470, 500Hz
	Bass Gain	0 … 12 dB
	Bass Slope	6,12,18 dB/Oct
Low	Low Frequency	20 ... 500 Hz
	Low Gain	−10.0 … +10.0 dB
	Low Width	0.8 ... 4 octaves
Mid	Mid Frequency	250 ... 6000 Hz
	Mid Gain	−10.0 … +10.0 dB
	Mid Width	0.8 ... 4 octaves
High	High Frequency	1.0 … 15.0 kHz
	High Gain	−10.0 … +10.0 dB
	High Width	0.8 ... 4 octaves
All	All Frequency	20 Hz … 20.0 kHz
	All Gain	−10.0 … +10.0 dB
	All Width	0.8 ... 4 octaves
HF Enhancer	High Frequency Enhancer	0 … 15 dB
	Sensitivity Trim	–10 … +10 dB
	Threshold Trim	0 … +20 dB
Subharmonic Injection	Subharmonic Injection	Off, −99.0 … 0 dB
	Subharmonic Cutoff Freq	90Hz, 120Hz
DJ Bass (5B)	DJ Bass Boost	Off, 1… +10 dB
Highpass Filter, Speech HP Filter	Highpass Filter Freq	Off, 20, 30, 40, 50, 60, 70, 80, 90, 100, 110, 120, 130, 140, 150, 170, 200 Hz
	Highpass Filter Slope	6, 12, 18, 24 dB / octave
LP Filter (EQ)	Lowpass Filter Freq	4, 5, 6, 7, 8, 9, 10, 11, 12, 13, 14, 15 kHz
	Lowpass Filter Slope	6, 12, 18, 24 dB / octave
LP Filter (steep)	Lowpass Filter	10, 11 ,12, 13, 14, 15, 20 kHz
Phase Rotate	Phase Rotate	In / Out

Table 3-8: Equalizer Controls

curves. The width (in octaves) is calibrated with reference to 10 dB boost. As you decrease the amount of EQ gain (or start to cut), the width in octaves will decrease. However, the "Q" will stay constant.

"Q" is a mathematical parameter that relates to how fast ringing damps out. (Technically, we are referring to the "Q" of the poles of the equalizer transfer function, which does not change as you adjust the amount of boost or cut.)

The curves in OPTIMOD-PCn's equalizer were created by a so-called "minimax" ("minimize the maximum error" or "equal-ripple") IIR digital approximation to the curves provided by the Orban 622B analog parametric equalizer. Therefore, unlike less sophisticated digital equalizers that use the "bilinear transformation" to generate EQ curves, the shapes of OPTIMOD-PCn's curves are not distorted at high frequencies.

Midrange Parametric Equalizer is a parametric equalizer whose boost and cut curves closely emulate those of an analog parametric equalizer with conventional bell-shaped curves.

Mid Freq determines the center frequency of the equalization, in Hertz. Range is 250-6000Hz.

Mid Gain determines the amount of peak boost or cut (in dB) over a ±10 dB range.

Mid Width determines the bandwidth of the equalization, in octaves. The range is 0.8-4.0 octaves. If you are unfamiliar with using a parametric equalizer, one octave is a good starting point.

With Five-Band presets, the audible effect of the midrange equalizer is closely associated with the amount of gain reduction in the midrange bands. With small amounts of gain reduction, it boosts power in the presence region. This can increase the loudness of such material substantially. As you increase the gain reduction in the midrange bands (by turning the MULTIBAND DRIVE control up), the MID GAIN control will have progressively less audible effect. The compressor for the midrange bands will tend to reduce the effect of the MID frequency boost (in an attempt to keep the gain constant) to prevent excessive stridency in program material that already has a great deal of presence power. Therefore, with large amounts of gain reduction, the density of the presence region energy will be increased more than will the level of energy in that region. Because the 3.7 kHz band compressor is partially coupled to the gain reduction in the 6.2 kHz band in most presets (as set by the B4>5 COUPLING control), tuning MID FREQ to 2-4 kHz and turning up the MID GAIN control will decrease energy in the 6.2 kHz band—you will be increasing the gain reduction in both the 3.7 kHz and 6.2 kHz bands. You may wish to compensate for this effect by turning up the BRILLIANCE control.

With Two-Band presets, the midrange equalizer will behave much more as you might expect because the two-band structure cannot automatically re-equalize midrange energy. Instead, increasing midrange energy will moderately increase the Master band's gain reduction.

Use the mid frequency equalizer with caution. Excessive presence boost tends to be audibly strident and fatiguing. Moreover, the sound quality, although loud, can be very irritating. We suggest a maximum of 3 dB boost, although 10 dB is achievable. In some of our factory music presets, we use a 3 dB boost at 2.6 kHz to bring vocals more up-front.

High Frequency Parametric Equalizer is an equalizer whose boost and cut curves closely emulate those of an analog parametric equalizer with conventional bell-shaped curves.

High Freq determines the center frequency of the equalization, in Hertz. The range is 1-15 kHz

High Gain determines the amount of peak boost or cut over a ±10 dB range.

High Width determines the bandwidth of the equalization, in octaves. The range is 0.8-4.0 octaves. If you are unfamiliar with using a parametric equalizer, one octave is a good starting point.

Excessive high frequency boost can exaggerate hiss and distortion in program material that is less than perfectly clean. We suggest no more than 4 dB boost as a practical maximum, unless source material is primarily from high-quality digital sources. In several of our presets, we use this equalizer to boost the upper presence band (4.4 kHz) slightly, leaving broadband HF boost to the BRILLIANCE and/or HF ENHANCE controls.

All Frequency Parametric Equalizer is similar to the other parametric sections but can be swept over the entire audio band, from 20 Hz to 20 kHz.

Brilliance controls the drive to Band 5. The Band 5 compressor/limiter dynamically controls this boost, protecting the final limiter from excessive HF drive. We recommend a maximum of 4 dB of Brilliance boost and most people will prefer substantially less.

DJ Bass (“DJ Bass Boost“) control determines the amount of bass boost produced on some male voices. In its default OFF position, it causes the gain reduction of the lowest frequency band to move quickly to the same gain reduction as its nearest neighbor when gated. This fights any tendency of the lowest frequency band to develop significantly more gain than its neighbor when processing voice because voice will activate the gate frequently. Each time it does so, it will reset the gain of the lowest frequency band so that the gains of the two bottom bands are equal and the response in this frequency range is flat. The result is natural-sounding bass on male voice. This is particularly desirable for most video programming.

If you like a larger-than-life, “chesty” sound on male voice, set this control away from OFF. When so set, gating causes the gain reduction of the lowest frequency band to move to the same gain reduction (minus a gain offset equal to the numerical setting of the control) as its nearest neighbor when gated. You can therefore set the maximum gain difference between the two low frequency bands, producing considerable dynamic bass boost on voice. This setting might be appropriate for news and sports.

The difference will never exceed the difference that would have otherwise occurred if the lowest frequency band were gated independently. If you are familiar with older Orban processors like OPTIMOD-FM 8200, this is the maximum amount of boost that would have occurred if you had set their DJ BASS BOOST controls to ON.

The amount of bass boost will be highly dependent on the fundamental frequency of a given voice. If the fundamental frequency is far above 100Hz, there will be little voice energy in the bottom band and little or no audio bass boost can occur even if the gain of the bottom band is higher than the gain of its neighbor. As the fundamental frequency moves lower, more of this energy leaks into the bottom band, and you hear more bass boost. If the fundamental frequency is very low (a rarity), there will be enough energy in the bottom band to force significant gain reduction, and you will hear less bass boost than if the fundamental frequency were a bit higher.

This control is only available in the Five-Band structure.

If the GATE THRESH (Gate Threshold) control is turned OFF, the DJ BASS boost setting is disabled.

The **High Frequency Enhancer** is a program-adaptive, 6 dB/octave shelving equalizer with a 4 kHz turnover frequency. It constantly monitors the ratio between high frequency and broadband energy and adjusts the amount of equalization in an attempt to make this ratio constant as the program material changes. It can therefore create a bright, present sound without over-equalizing material that is already bright.

High Frequency Enhancer is a mix control for the high frequency enhancer that sets the amount of enhanced high frequencies being mixed into the input and clamps the maximum enhancement to the control's setting.

Sensitivity Trim (v0.9.9 and higher) trims the ratio of high-frequency to wideband energy produced by HF Enhancer at a given setting of the HIGH FREQUENCY ENHANCER control. Higher settings increase the amount of enhancement. Default is 0 dB, which retains the behavior of the previous high frequency enhancer. This had only one control: HIGH FREQUENCY ENHANCER.

Threshold Trim (v0.9.9 and higher) trims the high frequency enhancer input level above which full HF enhancement occurs. Higher settings produce lower amounts of enhancement at low levels. Default is 0 dB, which produces the same behavior as previous versions of the HF enhancer. Note that the HF Enhancer is driven by the AGC, so its input level tracks the AGC's output level.

Subharmonic Injection, Subharmonic Cutoff Freq The subharmonic synthesizer generates subharmonics of fundamental frequencies in the 50-90 or 60-120 Hz range, as selected by the SUBHARMONIC CUTOFF FREQ control. The subharmonics are one octave below the frequencies from which they are generated (i.e., 25 to 45 Hz or 30-60 Hz) and track the levels of their generating frequencies. Subharmonic injection can be varied from 0 to 100% of the level of the generating frequency. 50% is a good starting point for popular music formats.

If input program material below 50 or 60 Hz is present, the subharmonic synthesizer automatically reduces the level of the synthesized subharmonics to prevent excess build-up of energy below 50 or 60 Hz (i.e. the amount of subharmonic injection will be lower than the setting of the SUBHARMONIC INJECTION control).

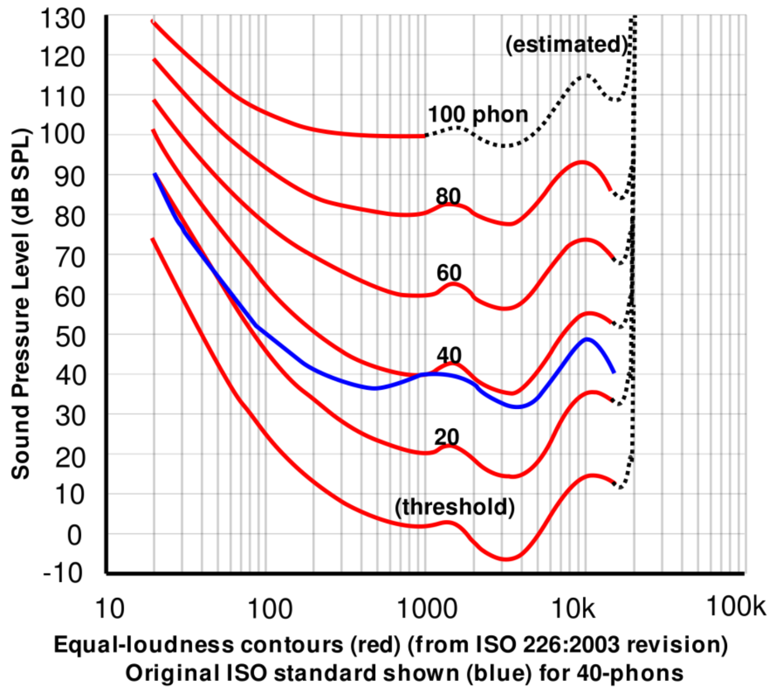


Figure 3-15: Equal Loudness Curves

To prevent introducing unnatural coloration in male speech, the subharmonic synthesizer is defeated when the Optimod's automatic speech/music detector detects speech. When you use the 120 Hz cutoff frequency (which can cause "thumps" behind low-pitched male speech), it is wise to ensure that raw speech is panned to the exact center, which is one important criterion that the speech/music detector uses to detect speech.

It is important to understand that material below 50 Hz takes up lots of peak level to produce significant loudness. Moreover, the close spacing of psychoacoustic "equal loudness" curves[3] below 50 Hz means small changes in amplitude lead to large change in subharmonic loudness. The unpredictability of receiver bass response means that the perceived loudness of subharmonics is highly receiver-dependent. Because of the amount of peak level they use up, subharmonics will always make a broadcast sound quieter for a given amount of processing arti-

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[3] The equal-loudness curves are popularly known as "Fletcher/Munson" curves, although these have been superseded by ISO 226:2003. The image was taken from en.wikipedia.org.

facts/distortion—they use up peak level that otherwise could be dedicated to audio to which the ear is more sensitive. (This is an inevitable effect of the equal-loudness curves.) For all these reasons, it is wise not to overdo subharmonic synthesis.

> Note that when you apply an L+R sinewave at a frequency between 50 to 90 Hz to the 1600's input, the 1600's speech/music detector will detect this as "speech," so no subharmonics will be produced. To test the subharmonic synthesizer with tone, there must be at least 2 dB of level difference between the left and right inputs. In other words, be sure that the 1600's speech/music detector is indicating "music" in the 1600 PC GUI.

<u>Highpass Filter, Speech Highpass Filter</u> determines if a sweepable 6, 12, 18 or 24 dB/octave highpass filter will be placed in-circuit before other processing. The 6, 12 and 18 dB/octave filters have a Butterworth response and the indicated frequency is where the filter is 3 dB down. The 24 dB/octave filter has a Chebychev response with 0.01 dB of passband ripple. The indicated highpass frequency of this filter is the passband edge of the filter. (For example, if the filter is set to 20 Hz, the high filter's response will be flat ±0.01 dB to 20 Hz.

The highpass filter is useful for reducing low frequency noise, particularly when OPTIMOD-PCn is being used for production or mastering.

Two control sets (each containing FREQUENCY and SLOPE) are available, one for music mode and one for speech mode. These can be switched via OPTIMOD-PCn's automatic speech/music detector.

> It is wise to make sure that the main and speech-mode highpass filter settings are the same unless you specifically want to apply additional highpass filtering to speech. Like all automatic speech/music detectors, the one in your Optimod can occasionally make mistakes, so it is also wise in mixed-format programming to set the filter no higher than 70 Hz. This way, if the Optimod mistakenly identifies music as speech (or decides that voice-over-music is speech), it will not cause an obvious dropout of bass frequencies.
>
> Note that an L=R test tone whose level changes depending on whether speech or music is detected can repeatedly toggle the speech/music detector between "speech" and "music" modes. This is because the sudden jump in tone level is "speech-like" and triggers the speech/music detector's syllabic detector. An example is a 100 Hz tone when the main highpass filter frequency is 50 Hz and the speech highpass filter frequency is 200 Hz. This issue never occurs with program material.

<u>Lowpass Filter (steep)</u> control sets the bandwidth (and therefore the amount of high frequency signal OPTIMOD-PCn passes) from 10 kHz to 20 kHz. The lowpass filter can replace any anti-aliasing filters in downstream equipment. Set the filter to 20 kHz (full bandwidth) for downstream equipment with sample rates of 44.1 or 48 kHz. Set the filter to 15 kHz for 32 kHz sample rate. For other sample rates, set the filter so that it is as close as possible to 45% of the sample rate without exceeding 45%.

> This setting is unique to the preset in which it resides. Regardless of its setting, OPTIMOD-PCn will not permit the system bandwidth to exceed

the bandwidth set by the MAX LOWPASS FILTER parameter located in the Configuration page of the I/O Mixer.

Lowpass Filter (gentle) is intended for program equalization, not anti-aliasing like the steep lowpass filter. It is intended mainly for mastering where the steep lowpass filter might cause audibly objectionable ringing. It is sweepable from 4 to 15 kHz with slopes of 6, 12, 18 or 24 dB/octave. The filter's shape is Butterworth, and the setting indicates the frequency where the filter is 3 dB down.

Phase Rotator determines if the phase rotator will be in-circuit. The purpose of the phase rotator is to make voice waveforms more symmetrical. Because it can slightly reduce the clarity and definition of program material, we recommend leaving it OUT unless program material is mainly speech, where it may result in cleaner sound because it can substantially reduce the amount of gain reduction that OPTIMOD-PCn's look-ahead limiter produces on speech waveforms.

Multiband Compressor Controls

Following the AGC is an equalization section, a five-band compressor, a dynamic single-ended noise reduction system, an output mixer (for the five bands), and a peak limiter.

When the input is noisy, you can sometimes reduce the noise by activating the single-ended noise reduction system. Functionally, the single-ended noise reduction

Multiband Controls	
Advanced Name	**Range**
Multiband Drive	0 ... 25
Multiband Gate Threshold	Off, −44 ... −15 dB
Downward Expander	Off, −18.0 … 12.0 dB
Speech Downward Expander	Off, −18.0 … 12.0 dB
B5 Downward Expander	Off, −18.0 … 12.0 dB
Downward Expander Stereo Coupling	On, Off
Crossover	Linear, Allpass
B1/B2 Crossover	100 Hz, 150 Hz, 200 Hz
B1/B2 Crossover Slope	Shallow, Steep
B1 MaxDeltGr	0 … 24 dB, Off
B2 MaxDeltGr	0 … 24 dB, Off
B3 MaxDeltGr	0 … 24 dB, Off
B4 MaxDeltGr	0 … 24 dB, Off
B5 MaxDeltGr	0 … 24 dB, Off
Loudness Threshold	Off, +10.0 … -14.0 dB
Loudness Attack	0…100%
Loudness Bass Couple	0…12 dB, Off
BS.1770 Threshold	0…6 dB, Off
Mono Bass	On, Off
Mono Bass Crossover	80, 100 Hz
Compressor Mode	Inline, Parallel
Compression Thresh Offset	0…–60 dB, 1 dB steps

Table 3-9: Multiband Controls

system combines a broadband downward expander with a program-dependent low-pass filter. This noise reduction can be valuable in reducing audible hiss, rumble, or ambient studio noise. We use it for the news and sports factory presets.

The Five-Band structure does not have a separate Loudness Controller because its Five-Band compressor automatically re-equalizes the spectral balance of various pieces of program material in a way that tends to make their loudness more consistent.

The Five-Band Structure's Full and Advanced Setup Controls

The tables below summarize the Five-band and Band Mix controls in the dynamics section. The AGC, Equalizer, Stereo Enhancer, and Clipper controls are common to both the Two-Band and Five-Band structures and are discussed in their own sections in Section 3.

Multiband Drive control adjusts the signal level going into the five-band compressor, and therefore determines the average amount of gain reduction in the five-band compressor. Range is 25dB. It works differently in INLINE and PARALLEL compressor modes. In INLINE mode, the control sets the drive into the multiband compression and thus the amount of gain reduction that it produces.

In parallel mode, the MULTIBAND DRIVE control is repurposed so that it sets the amount of compressor output that is added to the compressor's input. Its setting indicates the amount of amplification of quiet material that occurs when the compressor produces no gain reduction. It does not affect the amount of compressor gain reduction; set this with the COMPRESSION THRESH OFFSET control, which only works in parallel mode. See *Parallel Compression* on page 3-21.

In PARALLEL mode, it is a mix control located after the multiband compressor and sets how much of the compressor's output is added to its input signal before being passed to the next processing stage.

Adjust the MULTIBAND DRIVE control to your taste and programming requirements. Used lightly with a slow or medium release time, the Five-Band compressor produces an open, re-equalized sound that is appropriate for most video programming. The Five-Band compressor can increase audio density when operated at a fast or medium-fast release because it acts more and more like a fast limiter (not a compressor) as the release time is shortened. With fast and medium-fast release times, density also increases when you increase the drive level into the Five-Band compressor because these faster release times produce more limiting action. Increasing density can make loud sounds seem louder, but can also result in an unattractive busier, flatter, or denser sound. It is very important to be aware of the many negative subjective side effects of excessive density when setting controls that affect the density of the processed sound.

Because OPTIMOD-PCn's AGC algorithm uses sophisticated window gating, it is preferable to make the AGC do most of the gain riding (instead of the five-band compressor), because the AGC can ride gain quickly without adding excessive density to program material that is already well controlled. The five-band compressor is typically used with 5 to 10 dB of average gain reduction so it can perform automatic re-equalization of material that the AGC has already controlled without adding excessive density to the audio or re-equalizing to an unnatural extent.

The MULTIBAND DRIVE interacts with the MULTIBAND RELEASE. With slower release time settings, increasing the MULTIBAND DRIVE control scarcely affects density. Instead, the primary danger is that the excessive drive will cause noise to be increased excessively when the program material becomes quiet. You can minimize this effect by activating the single-ended noise reduction and/or by carefully setting the MULTIBAND GATE THRESHOLD control to freeze the gain when the input gets quiet.

When the release time of the Five-Band compressor is set towards FAST, the setting of the MULTIBAND DRIVE control becomes much more critical to sound quality because density increases as the control is turned up. Listen carefully as you adjust it. With these fast release times, there is a point beyond which increasing the Five-Band compressor drive will no longer yield more loudness, and will simply degrade the punch and definition of the sound. Instead, let the AGC do most of the work.

Because excessive loudness is an irritant in sound for picture, there is almost never any reason to push processing to the point where it degrades the audio. We recommend no more than 10dB gain reduction as shown on the meters for Band 3. More than 10dB, particularly with the fast release time, will often create a wall of sound effect that many find fatiguing.

To avoid excessive density with fast Five-Band release time, we recommend using no more than 5dB gain reduction in band 3, compensating for any lost loudness by speeding up the AGC RELEASE instead.

Multiband Release; Speech Multiband Release control can be switched to any of seven settings. To understand how to adjust this control for video programming, please

Multiband Attack/Release/Threshold	
Advanced Name	**Range**
Multiband Release	Slow, Slow2, Med, Med2, MFast, MFast2, Fast
Speech Multiband Release	Slow, Slow2, Med, Med2, MFast, MFast2, Fast
Loudness Threshold	Off, 0.0 … -24.0 dB
Loudness Attack	0…100%
Loudness Bass Couple	0…12 dB, Off
Bx Compression Threshold	−16.00 … 0.0, Off
Bx Speech Compression Threshold	−16.00 … 0.0, Off
Bx Attack	4.0 … 50.0 ms, Off
Speech Bx Attack	4.0 … 50.0 ms, Off
Bx Limiter Attack	0 … 100%
Bx Delta Release	−6 … 6
Bx Knee	1…50 dB
Bx Compression Ratio	1:1… ∞:1
Bx Breakpoint	1…50 dB
Transient Enhance	0…10 ms

Table 3-10: Multiband Attack / Release Controls

see the discussion above under MB DRIVE.

The SPEECH MB RELEASE control overrides the MB RELEASE control when OPTIMOD-PCn automatically detects speech (page 3-7). You may wish to set the SPEECH MB RELEASE control faster for speech (to maximize smoothness and uniformity) and slower on music (to prevent excessive build-up of density).

Figure 3-12 on page 3-28 shows the effect of compressor attack and release on the amplitude of the audio waveform.

<u>Compression Threshold; Speech Compression Threshold</u> controls set the compression threshold for music and speech in each band (following OPTIMOD-PCn's automatic speech/music discriminator), in units of dB. We recommend making small changes around the factory settings to preserve the internal headroom built into the processing chain. These controls will affect the spectral balance of the processing above threshold, but are also risky because they can significantly affect the amount of distortion produced by the back-end clipping system.

You can use these controls to set independent frequency balances for music and speech (page 3-7).

<u>MB Gate</u> ("Multiband Gate Threshold") control determines the lowest input level that will be recognized as program by OPTIMOD-PCn; lower levels are considered to be noise or background sounds and cause the AGC or multiband compressor to gate, effectively freezing gain to prevent noise breathing.

There are two independent gating circuits in OPTIMOD-PCn. The first affects the AGC and the second affects the five-band compressor. Each has its own threshold control.

The multiband silence gate causes the gain reduction in bands 2 and 3 of the five-band compressor to move quickly to the average gain reduction occurring in those bands when the gate first turns on. This prevents obvious midrange coloration under gated conditions, because bands 2 and 3 have the same gain.

The gate also independently freezes the gain of the two highest frequency bands (forcing the gain of the highest frequency band to be identical to its lower neighbor), and independently sets the gain of the lowest frequency band according to the setting of the DJ BASS boost control (in the Equalization screen). Thus, without introducing obvious coloration, the gating smoothly preserves the average overall frequency response "tilt" of the five-band compressor, broadly maintaining the "automatic equalization" curve it generates for a given piece of program material.

> If the MB GATE control is turned OFF, the DJ BASS control (in the Equalization screen) is disabled.

<u>Crossover</u> sets the structure of the five-band crossover to ALLPASS or LINEAR.

In LINEAR mode, the delay through the five-band compressor is essentially constant at all frequencies. Accordingly, each crossover band is down 6 dB at the crossover frequency.

In ALLPASS mode, the magnitude response through the crossover is flat when all bands have the same amount of gain reduction, but the five-band compressor applies an overall phase rotation to the sound—different frequencies are delayed by different amounts of time. Each crossover band is down 3 dB at the crossover frequency.

In general, LINEAR sounds more “precise” than ALLPASS and preserves transient detail better. However, ALLPASS can sound “fuller” or “bigger” than LINEAR. The audible difference between the two is often subtle and neither is “correct”; choosing a crossover type is a matter of preference. Most loudspeakers, including many highly regarded models, use allpass crossovers, and given this fact, it is important not to assume that LINEAR is always correct just because it introduces no phase distortion.

Figure 3-8 on page 3-19 is a conceptual depiction of the frequency response of the crossover filters.

<u>B1/B2 Crossover</u> (Band 1 to Band 2 Crossover Frequency) sets the crossover frequency between bands 1 and 2 to either 100 Hz or 200 Hz. It significantly affects the bass texture, and the best way to understand the differences between the two crossover frequencies is to listen.

<u>B1/B2 Crossover Slope</u> (SHALLOW, STEEP) determines the selectivity of the filter used to separate the frequencies applied to the band 1 and band 2 crossover sidechains, which compute the gain applied to the audio in each frequency band. The control does not affect the shape of the crossover in the audio path, only in the sidechain, where it determines how much the band 1 and band 2 gain reductions are affected by frequencies outside the main passband of a given band. For example, when this control is set to STEEP and the B1/B2 CROSSOVER control is set to 200 Hz, a strong signal at 100 Hz (which is outside the main frequency range covered by band 2) will produce less gain reduction in band 2 than it would if the control was set to SHALLOW.

Subjectively, a setting of STEEP produces a more consistent bass texture between various program elements than does a setting of SHALLOW. STEEP is the preferred choice if you want consistent, punchy bass texture. On the other hand, SHALLOW preserves more of the original bass coloration of the source because it produces less gain reduction difference between band 1 and band 2 than does STEEP.

<u>Mono Bass</u> causes the two input channels to be blended to mono below a frequency set by the MONO BASS CROSSOVER control. This function is implemented via a linear-phase highpass filter applied to the stereo difference channel (L−R). A compensating delay in the stereo sum channel makes the transition between subjectively perfect separation and mono as narrowband as possible.

If there is significant bass energy in the L−R channel, this process will reduce the overall bass heard by the listener. Because the process precedes the multiband compressor, this will tend to restore this loss of bass by producing less gain reduction in band 1. This will be most effective if the B1/B2 CROSSOVER SLOPE control is set to STEEP and you use 100 Hz B1/B2 CROSSOVER.

Sometimes, it is useful to use the bass shelving equalizer in the EQ section to produce a bit of bass boost to compensate statically for bass loss caused by the mono bass process.

Mono Bass Crossover (see MONO BASS above)

MB Downward Expander ("Multiband Downward Expander Threshold") determines the level below which the single-ended noise reduction system's downward expander begins to decrease system gain and below which the high frequencies begin to become low-pass filtered to reduce perceived noise. There are three controls: the MB DOWN EXPANDER and SPEECH MB DOWN EXPANDER controls set the expansion threshold in Bands 1-4 for Music and Speech modes, while the B5 DOWN EXPANDER DELTA THRESHOLD control offsets the expansion threshold in Band 5 with respect to the active MB DOWN EXPANDER threshold for both Speech and Music modes. Activate the single-ended dynamic noise reduction by setting these controls to a setting other than OFF.

The single-ended noise reduction system combines a broadband downward expander with a program-dependent low-pass filter. These functions are implemented by causing extra gain reduction in the multiband compressor. You can see the effect of this extra gain reduction on the gain reduction meters. The maximum expander gain reduction achievable is constrained to 6 dB in bands 2-5 and 18 dB in band 1, as this range often contains hum and/or rumble that can benefit from extra noise reduction.

Ordinarily, the gating on the AGC and multiband limiter will prevent objectionable build-up of noise and you will want to use the single-ended noise reduction only on unusually noisy program material. Modern commercial recordings will almost never need it. We expect that its main use will be in talk-oriented programming, including sports.

Please note that it is impossible to design such a system to handle all program material without audible side effects. You will get best results if you set the MB DOWN EXPANDER control of the noise reduction system to complement the program material you are processing. The MB DOWN EXPANDER should be set higher when the input is noisy and lower when the input is relatively quiet. The best way to adjust the MB DOWN EXPANDER control is to start with the control set very high. Reduce the control setting while watching the gain reduction meters. Eventually, you will see the gain increase in sync with the program. Go further until you begin to hear noise modulation—a puffing or breathing sound (the input noise) in sync with the input program material. Set the MB DOWN EXPANDER control higher until you can no longer hear the noise modulation. This is the best setting.

Obviously, the correct setting will be different for a sporting event than for classical music. It may be wise to define several presets with different settings of the MB

DOWN EXPANDER control and to recall the preset that complements the program material of the moment.

Note also that it is virtually impossible to achieve undetectable dynamic noise reduction of program material that is extremely noisy to begin with, because the program never masks the noise. It is probably wiser to defeat the dynamic noise reduction with this sort of material (traffic reports from helicopters and the like) to avoid objectionable side effects. You must let your ears guide you.

Band 5 is particularly critical for noise reduction because much of the Downward Expander's utility lies in hiss reduction. Hiss has most of its energy in band 5, while program material typically has less energy in this band, so the B5 DOWN EXPANDER control's setting is critical to removing hiss while minimizing removal of desired program energy.

Band 5 is uncoupled from the lower bands so the band 5 downward expander can produce less gain reduction than other bands. This can help prevent loss of desired high frequency material in the program.

B3>B4 CPL ("Band 3>4 Coupling") control determines the extent to which the gains of band 4 (centered at 3.7 kHz) and 5 (above 6.2 kHz) are determined by and follow the gain of band 3 (centered at 1 kHz). Set towards 100% (fully coupled) this control reduces the amount of dynamic upper midrange boost, preventing unnatural upper midrange boost. The gain of band 5 is further affected by the B4>B5 CPL control.

Excessive HF energy is one cause of audibly objectionable artifacts in low bitrate codecs. The B3>B4 CPL AND B4>B5 CPL controls can be very useful in reducing such artifacts: Setting them for large amounts of coupling will minimize OPTIMOD-PCn's abil-

Band Mix	
Advanced Name	**Range**
B2>B1 Coupling	0 ... 100 %
B2>B3 Coupling	0 ... 100 %
B3>B2 Coupling	0 … 100 %
B3>B4 Coupling	0 ... 100 %
B4>B5 Coupling	0 ... 100 %
B1 Output Mix	−6.0 … +6.0
B2 Output Mix	−6.0 … +6.0
B3 Output Mix	−6.0 … +6.0
B4 Output Mix	−6.0 … +6.0
B5 Output Mix	−6.0 … +6.0
B1 On/Off	On, Off
B2 On/Off	On, Off
B3 On/Off	On, Off
B4 On/Off	On, Off
B5 On/Off	On, Off

Table 3-11: MB Band Mix Controls

ity to increase high frequency energy dynamically.

B4>B5 CPL ("Band 4>5 Coupling") controls the extent to which the gain of band 5 (6.2 kHz and above) is determined by and follows the gain of band 4.

The sum of the high frequency limiter control signal and the output of the B4>B5 CPL control determines the gain reduction in band 5. The B4>B5 CPL control receives the independent left and right band 4 gain control signal. Range is 0 to 100% coupling.

B3>B2 Coupling and B2>B3 Coupling controls determine the extent to which the gains of bands 2 and 3 track each other.

When combined with the other coupling controls, these controls can adjust the five-band processing to be anything from fully independent operation to quasi-wideband processing.

B2>B1 Coupling control determines the extent to which the gain of band 1 (below 100Hz or 200Hz, depending on crossover setting) is determined by and follows the gain of band 2 (centered at 400Hz). Set towards 100% (fully coupled), it reduces the amount of dynamic bass boost, preventing unnatural bass boost. Set towards 0% (independent), it permits frequencies below 100Hz (the "slam" region) to have maximum impact in modern rock, urban, dance, rap, and other music where bass punch is crucial. Accordingly, it can be useful in music video oriented formats.

Bx Output Mix Because these controls mix *after* the band compressors, they do not affect the compressors' gain reductions and can be used as a graphic equalizer to fine-tune the spectral balance of the program material over a ±6 dB range.

Their range has been purposely limited because the only gain control element after these controls is the look-ahead limiter, which can produce pumping or distortion if overdriven. The thresholds of the individual compressors have been tuned to prevent audible distortion with almost any program material. Large changes in the frequency balance of the compressor outputs will change this tuning, leaving OPTIMOD-PCn more vulnerable to unexpected audible distortion with certain program material.

You can also get a similar effect by adjusting the compression threshold of the individual bands. This is comparably risky with reference to look-ahead limiter overload, but unlike the MB BAND MIX controls, the threshold adjustments do not affect the frequency response when a given band is below threshold and is thus producing no gain reduction.

B1-B5 On/Off switches allow you to mute any combination of bands in the five-band compressor and permit you to "solo" any individual band. This can be useful when you are designing new user presets.

B1-B5 Attack (Time); Speech B1-B5 Attack controls set the speed with which the gain reduction in each band responds to level changes at the input to a given band's compressor for music and speech respectively, following OPTIMOD-PCn's automatic speech/music detector. These controls are risky and difficult to adjust appropriately. They affect the sound of the processor in many subtle ways. The main trade-off is "punch" (achieved with slower attack times) versus distortion and/or pumping produced in the look-ahead limiter (because slower attack times increase overshoots that the look-ahead limit must eliminate). The results are strongly program-dependent and must be verified with listening tests to a wide variety of program material.

Because there are separate controls for music and speech (page 3-7), you can set attack times faster for speech (to minimize look-ahead limiter artifacts) and slower for music (to maximize punch and transient definition).

The ATTACK time controls are calibrated in arbitrary units that very approximately correspond to milliseconds. Higher numbers correspond to slower attacks.

Limiter Attack controls allow you to set the limiter attack anywhere from 0 to 100% of normal in the Five-Band compressors, each of whose gain reduction has a fast-release (limiter) and slow-release (compressor) component. Because the limiter and compressor characteristics interact, you will usually get best audible results when you set these controls in the range of 70% to 100%. Below 70%, you will usually hear gain pumping because the compressor function is trying to create some of the gain reduction that the faster limiting function would have otherwise achieved. If you hear pumping in a band and you still wish to adjust the limiter attack to a low setting, you can sometimes ameliorate or eliminate the pumping by slowing down the compressor attack time in that band.

These controls have nothing to do with the final peak limiter.

Delta Release controls are differential controls. They allow you to vary the release time in any band of the Five-Band compressor/limiter by setting an offset between the MULTIBAND RELEASE setting and the actual release time you achieve in a given band. For example, if you set the MULTIBAND RELEASE control to medium-fast and the BAND 3 DELTA GR control to −2, then the band 3 release time will be the same as if you had set the MULTIBAND RELEASE control to medium and set the BAND 3 DELTA GR control to 0. Thus, your settings automatically track any changes you make in the MULTIBAND RELEASE control. In our example, the release time in band 3 will always be two "click stops" slower than the setting of the MULTIBAND RELEASE control.

If your setting of a given DELTA RELEASE control would otherwise create a release slower than "slow" or faster than "fast" (the two end-stops of the MULTIBAND RELEASE control), the band in question will instead set its release time at the appropriate end-stop.

Band 1-5 Max Delta GR controls set the maximum permitted gain difference between the left and right channels for each band in the multiband limiter. The five-band processing chain uses a full dual-mono architecture, so the channels can be operated anywhere from fully coupled to independent. We recommend operating band 1-4 fully coupled (BAND 1-4 MAX DELTA GR = 0) for best stereo image stability. However, audio-processing experts may want to experiment with lesser amounts of coupling to achieve a wider, "fatter" stereo image at the cost of some image instability.

B5 MAX DELTA GR is set OFF most factory presets. This permits band 5 to be used as a fast-operating high frequency limiter that works independently on the left and right channels. This prevents gain reduction in one channel from causing audible spectral modulation on the other channel. However, the additional stereo difference channel energy created by independent operation can adversely affect certain low bitrate codecs (like WMA). It is wise to do careful listening tests through the codec to determine if it sounds better with B5 MAX DELTA GR = 0 dB.

B1-B5 Compression Ratio sets the compression ratio of a given band at its thresholds of compression. Beyond threshold, the ratio increases with increased gain reduction until it becomes ∞:1 at the amount of gain reduction (in dB) set by the B[x] COMPRESSOR KNEE control. When you adjust these controls, the thresholds of the multiband compressors change automatically so that the total amount of gain reduction stays approximately the same. (This automatic adjustment is internal to the

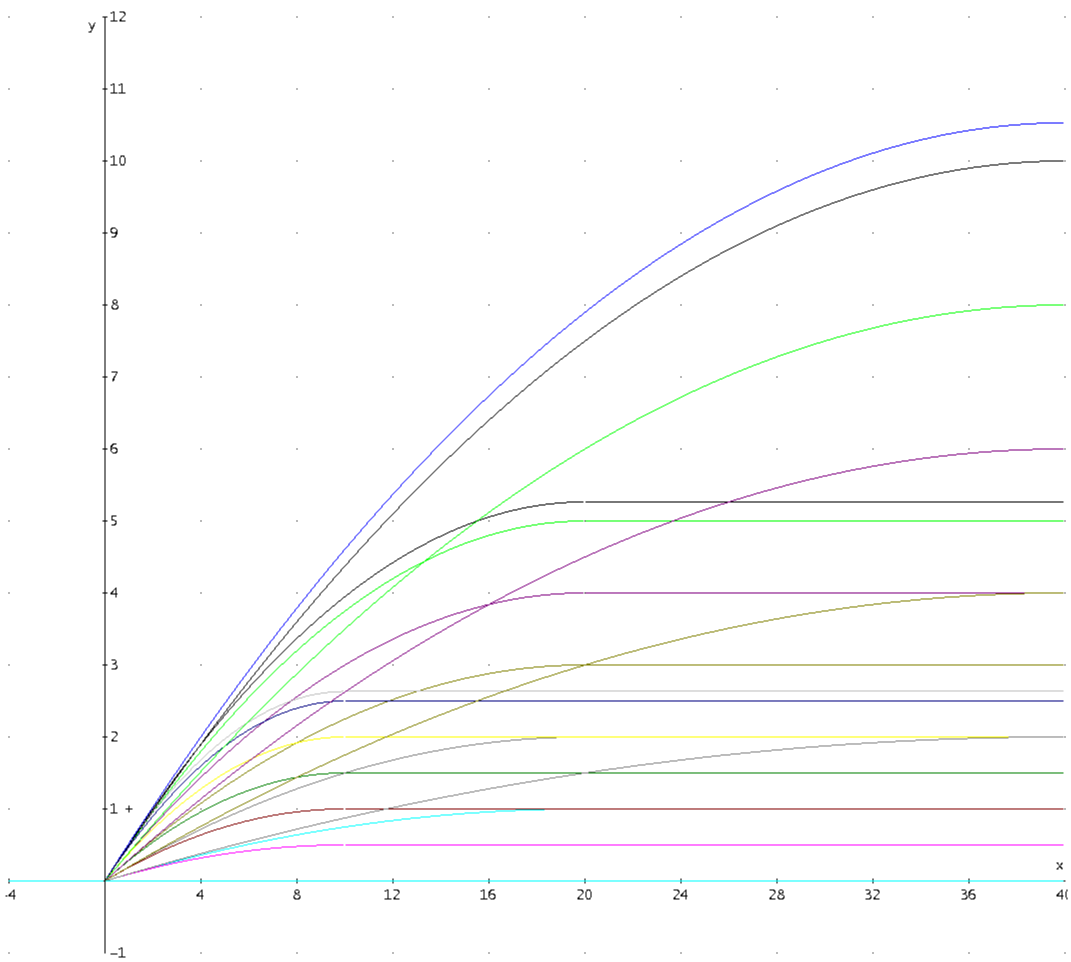


Figure 3-16: Output level in dB (y) for a given input level in dB (x) at various settings of the KNEE and RATIO control

1600's DSP; the MB THRESH controls' displayed settings do not show it.)

To achieve a classic soft knee characteristic, set the COMPRESSION RATIO control to 1:1 and set the COMPRESSION KNEE control to the gain reduction in dB at which you wish the compression ratio to level off to ∞:1. The maximum setting produces the softest knee. Setting the KNEE to 0 dB produces a classic hard knee curve with ∞:1 compression ratio regardless of the setting of the corresponding COMPRESSION RATIO control.

See Figure 3-16 on page 3-71 for the curves of output level vs. input level for various settings of the KNEE and RATIO controls. See Figure 3-9 on page 3-22 for the input/output curve in parallel compression mode.

Compression Knee (see COMPRESSION RATIO above).

Compression Breakpoint The release rate (measured in dB/second) in the 1600's compressors is constant when the gain reduction is higher than the control's setting, and exponential when the gain reduction is lower than the control's setting.

When the release is exponential, the release rate is proportional to the amount of gain reduction..

Compression-induced audio density remains constant when the gain reduction is above the BREAKPOINT setting. When the gain reduction is below the BREAKPOINT setting, density decreases proportionally to the amount of gain reduction.

For example, if the BREAKPOINT is set to 10 DB, the release rate (in dB/second) will be constant when the gain reduction is above 10 dB. Between 10 dB and 0 dB gain reduction, the release rate will slow down more and more.

The calibration of the BREAKPOINT controls is only accurate when KNEE = 0 dB and/or RATIO = infinity:1 — i.e., when the compression ratio is essentially infinite. When the ratio is less than infinite, the effective breakpoint of the compressor will be lower than COMPRESSOR BREAKPOINT setting.

The main use of the COMPRESSOR BREAKPOINT control is to prevent the compressor from objectionably increasing audio density when using low compression ratios and a significant amount of gain reduction—for example, 10 dB. The BREAKPOINT control is best adjusted by ear. If you find that density increases too much as gain reduction increases, lower the COMPRESSOR BREAKPOINT control's setting. If you want more density at high amounts of gain reduction, increase the COMPRESSOR BREAKPOINT control's setting. 10 dB is a good starting point for setting this control.

Loudness Threshold sets the maximum subjective loudness allowed by the CBS Loudness Controller with reference to the input of the 1600's MB look-ahead limiter. (See *Loudness Control* on page 3-24.)

Loudness Controller Attack: See *Loudness Control* on page 3- 24.

Loudness Controller Bass Couple: See *Loudness Control* on page 3- 24.

<u>**BS.1770 Threshold**</u>: See *BS.1770 Safety Limiter* on page 3-26.

<u>**Compressor Mode:**</u> See *Parallel Compression* on page 3-21.

<u>**Compression Thresh Offset:**</u> is active only in parallel compression mode. See *Parallel Compression* on page 3-21.

Peak Limiter

Regardless of whether MX mode is ON or OFF, a bass pre-limiter stage controls low frequency peaks ahead of the main peak limiter, while a "true-peak"-aware look-ahead limiter (equivalent to the look-ahead limiter in Orban products like OPTIMOD-PC 1101) implements final peak control. This limiter typically constrains the true peak overshoot (i.e., the maximum peak value that would appear after an ideal digital-to-analog converter) to 0.2 dB or less. (See *Figure 3-10: Peak Limiter Output vs. Input* on page 3-23.)

> OPTIMOD-PCn's OUTPUT meter shows peak levels based on audio sample values. To assess the accuracy of OPTIMOD-PCn's true peak limiting, you can use the free Orban Loudness Meter, available for download from www.orban.com/meter. This displays the "Reconstructed Peak" level, which is the same as "True Peak" level.

<u>**Bass Clip Threshold**</u> sets the threshold of the bass pre-limiter with respect to the threshold of the final peak limiter. The BASS LIMITER gain reduction meters show the amount of bass clipping or limiting in each audio channel.

The SPEECH BASS CLIP THRESHOLD control overrides the BASS CLIP THRESHOLD control when OPTIMOD-PCn automatically detects speech (page 3-7).

<u>**Bass Pre-Limit Mode**</u> sets the operation of the bass pre-limiter to HARD, MEDIUM, or SOFT. In surround instances this control does not work when the MX limiter is OFF; in this case the bass clip mode is always MEDIUM.

- HARD produces the most midbass harmonic enhancement.

  This can be useful if you want maximum bass punch, because this setting allows bass transients (like kick drums) to make square waves. The peak level of the fundamental component of a square wave is 2.1 dB *higher* than the peak level of the flat top in the square wave. Therefore, this allows you to get low bass that is actually higher than 100% modulation (or 0 dBFS)—the phase of the harmonics produced by the clipping works to reduce the peak level of the overall program.

  The squarewaves produced by this clipper are lowpass-filtered to reduce the audibility of the higher clipper-generated harmonics. Nevertheless, the downside is that material with sustained bass (including speech) can sound somewhat less clean than it will with the MEDIUM or SOFT settings.

  > The HARD clip algorithm used in 1600 is improved compared to that used in older Optimods. It uses a different algorithm that is less likely to cause

audible distortion, so HARD bass clipping in the 1600 is more useful than it was in older Optimods.

- MEDIUM uses more sophisticated signal processing than HARD to reduce distortion substantially.
- SOFT uses the most sophisticated look-ahead signal processing to reduce distortion further.

The tradeoff of using MEDIUM or SOFT bass clipping is less bass punch compared to HARD.

MB Final Limit Drive control adjusts the level of the audio driving the peak limiter, thereby adjusting the peak-to-average ratio of the processed audio. The MB FINAL LIMIT DRIVE control primarily determines the loudness/distortion trade-off.

Turning up the MB FINAL LIMIT DRIVE control drives the look-ahead limiter harder, reducing the peak-to-average ratio, and increasing the loudness of the output. When the amount of limiting is increased, the audible intermodulation distortion caused by limiting increases, even though special algorithms minimize the increase compared to less sophisticated designs. Lower settings reduce loudness, of course, but result in a cleaner sound.

The MX peak limiter uses a variety of tactics to adapt its operation intelligently to the program material applied to it. Compared to the look-ahead peak limiter in non-MX mode, the MX peak limiter produces a different set of artifacts when overdriven. Depending on how you set the other controls, these artifacts may include loss of bass punch, harsh clipper-like distortion, "soft" IM distortion, and/or excessive density that can cause fatigue. If you intend to make adjustments to the FINAL LIMIT DRIVE control, it is wise to familiarize yourself with these artifacts by purposely overdriving the peak limiter via the MB FINAL LIMIT DRIVE control and listening to what happens with different types of program material.

When using preemphasis, it is usually necessary to turn down the MB FINAL LIMIT

Peak Limiter Controls	
Name	**Range**
Bass Clip Shape	0.0 … 10.0
Bass Clip Threshold	−6.0 … +6.00, Off
Bass Limiting	0…−6
BassPre-Limit Mode (all modes except surround MX-OFF)	Soft, Medium, Hard
Bass Pre-Limiting (all modes except surround MX-OFF)	0 … 1
Distortion Control (MX mode only)	0 … 1
HF Distortion Control (MX Mode only)	
Final Limit Drive	−10.0 … +12.0
MX Limiter	On, Off
MX Limiter Threshold (all modes except surround MX-OFF)	−6.0 … +6.00, Off
MX Overshoot Limit Mode (MX mode only)	Hard, Soft
Speech Bass Clip Threshold	−6.0 … +6.00, Off
Transient Enhance	0…10 ms

Table 3-12: Peak Limiter Controls

DRIVE control to prevent the peak limiter from causing objectionable artifacts.

> You may find it illuminating to recall several Factory Presets, adjust LESS-MORE to several points in its range, and then open the Full Control screen to examine the trade-offs between the release time and FINAL LIMIT drive made by the factory programmers. However, note that all Factory Presets were created to complement FLAT preemphasis. As explained above, you must turn down the FINAL LIMIT DRIVE control when using preemphasis.

Transient Enhance is mainly useful in mastering. This control allows you to insert an audio delay in the sidechain of the five-band compressor. By delaying the gain control signal, this allows attack transients to pass through the multiband compressor uncompressed, which can increase punch. There is a tradeoff between this control and the activity of the look-ahead limiter, which will have to eliminate attack transients exceeding the look-ahead limiter's threshold. For any material, there will be an optimum setting for the TRANSIENT ENHANCE control that provides the most punch without triggering peak limiter artifacts.

MX Technology and Non-MX Stereo/Mono Peak Limiter Controls

MX Limiter When MX LIMITER is ON, additional peak control occurs between the bass pre-limiter and the look-ahead limiter, and the look-ahead true-peak limiter is used very lightly for final overshoot control only. When the peak limiter section is being driven heavily, the MX peak limiter, which uses a psychoacoustic model, provides decreased audible distortion, more transient punch, a degree of "exciter"-like harmonic enhancement that you can adjust with the HF DISTORTION CONTROL control, and a crisper, more open high frequency texture.

When MX LIMITER is OFF in stereo and mono instances, the MX pre-processing is still active to reduce the drive into the final look-ahead limiter and control potential IM distortion. Surround instances omit the pre-processing in non-MX mode. Non-MX mode is more likely to cause audible gain pumping when driven heavily.

The MX limiter has two downsides: its sophisticated processing increases the amount of CPU power required compared to non-MX mode, and it adds up to several hundred milliseconds of input-to-output delay. The delay is required so that the MX limiter has time to make intelligent decisions about how to process. In stereo and mono instances, non-MX mode has the same delay as non-MX SOFT mode; in surround mode, the delay is lower in non-MX mode.

Defeating the MX limiter can allow more instances of processing to be run on the CPU.

When the target loudness is below −12 LUFS (as is typical for material "mastered for iTunes®"), the MX limiter is unlikely to provide significant audible advantages because the overall peak limiter section is not being driven hard, so the non-MX look-ahead limiter (which is itself quite capable) is unlikely to produce audible artifacts when used alone. If you want to minimize CPU usage, it is wise to perform listening tests to see if the MX limiter provides audible advantages with your program material.

In mastering applications, on the other hand, neither CPU usage nor delay are usually relevant, so you may wish to save time by always activating the MX limiter because it will very seldom degrade the sound compared to not using it.

When the target loudness is below −22 LUFS (as is typical is sound-for-picture applications), the MX limiter never provides audible advantages because at this target loudness, the peak limiter rarely produces any gain reduction at all.

The only exception to these recommendations is when preemphasis is enabled. When you use OPTIMOD-PCn in a preemphasized mode (step 4.C) on page 2-18), this inserts a frequency-dependent high frequency boost before peak limiter. This boost can be as large as 20 dB at 20 kHz. The peak limiter handles this boost far more smoothly when the MX limiter is active, so we strongly advise always using the MX limiter when you use preemphasis.

The three controls described below are active except in surround instances when the MX limiter is OFF. They allow you to trade off loudness, distortion, and bass energy. Increasing bass increases the likelihood that audible IM distortion between bass and other program elements will occur. Depending on the program material you are processing, you may prefer to have a cleaner sound (with less bass) or a sound with more distortion but punchier bass. We offer controls, explained below, that allow you to make this tradeoff.

MX Limiter Threshold sets the threshold of the peak limiter with respect to the threshold of the final look-ahead limiter, used for overshoot control. This control is normally set to 0 dB. When the peak limiter is driven very hard (as determined by the setting of the MB FINAL LIMIT DRIVE control), setting the MB LIMITER THRESHOLD slightly below 0 dB can reduce gain pumping artifacts that would otherwise be caused by the final look-ahead limiter.

Bass Pre-Limiting / Speech Bass Pre-Limiting The bass pre-limiter can intelligently reduce the bass applied to the main peak limiter to reduce or prevent audible IM distortion. It does so when the pre-limiter's analysis of the program material indicates that this action is needed to prevent or minimize audible IM distortion between the bass (125 Hz and below) and other program elements in the main peak limiter. The BASS PRE-LIMITING control allows you to specify the maximum amount of bass reduction that can occur. Lower settings increase bass punch but do not protect against IM distortion as effectively as higher settings do.

There are two controls, one for Speech mode and one for Music mode, allowing you to have separate settings depending on whether OPTIMOD-PCn automatically detects speech or music input.

Bass Limiting Like the bass pre-limiter, the main peak limiter can automatically reduce bass when it detects potentially audible IM distortion. The BASS LIMITING control allows you to limit the amount of potential bass reduction at the expense of a possible increase in IM distortion. The scale shows the maximum amount of dynamic bass cut that can be produced, in dB.

The three controls below are available only in MX mode.

Distortion Control This control determines the amount of audible distortion that the main peak limiter is permitted to create. Higher settings can increase loudness and punch at the expense of audible clipping distortion. Lower settings are cleaner but may reduce punch and loudness. We prefer it at 0, which is its cleanest setting. All MX factory presets use this setting.

The best way to familiarize yourself with the effects of this control is by listening extensively to different types of program material while experimenting with different settings of the control. Because the MX peak limiter uses advanced algorithms that differ from those used by past Orban processors, the loudness/distortion/brightness/punch tradeoffs are also different and it is worthwhile to take the time to get a feel for the MX limiter's capabilities.

HF Distortion Control determines the amount of program-adaptive harmonic enhancement that the MX limiter is permitted to create around 6 kHz. "1.0" gives the same amount of harmonic enhancement as previous versions of the software.

Turning down the control produces a cleaner sound that is less bright, but may have more "peak limiter" sound as the limiter's internal overshoot compensator works harder.

MX Overshoot Limit Mode (HARD, SOFT) determines the algorithm used to eliminate overshoots produced by the main part of the MX processing. HARD uses a complex, computationally intensive process that shares certain characteristics with clipping, while SOFT uses a look-ahead limiter. In general, low bitrate codecs will handle SOFT mode better than HARD mode.

HARD produces punchier transients and is often preferred when the absolute maximum loudness is desired. It adds about 200 ms of delay compared to SOFT, and this is easy to hear when one is switching back and forth between HARD and SOFT modes. (Switching between modes produces audible discontinuities and should not be done "on-air.")

For applications where the MX limiter is not driven to its limits, SOFT may be preferred because it has less distortion and loads the CPU significantly less. As always, you should choose based on program material and processing goals. Subjectively, it might be said that HARD is more "rock 'n' roll," while SOFT is more "polite."

HARD mode is very CPU-intensive and can overload some computers. See the **Warning** on page 3-35.

Note: Because the 1600 PCn software is not licensed as a processor for FM radio, HARD mode is locked out when the pre-emphasis is set to 50 μs or 75 μs.

Phase Corrector Controls

These controls are located in the LESS-MORE tab. See *Left/Right Phase Skew Correction* on page 3-14. Because the phase corrector is automatically bypassed when the

Optimod MODE is AUTO or UPMIX, you must use the Optimix delay corrector if you wish to phase-correct an upmix.

Phase Corrector Controls	
Name	**Range**
Phase Corrector	In, Out
Phase Corrector Crossover	62.5…796 Hz

Table 3-13: Phase Corrector Controls

Downmix Controls

[Downmix #2 is not available in v 0.9.10 software.]

Except as noted, these controls are located in the DOWNMIX tab.

Loudness Threshold sets the maximum subjective loudness allowed by the CBS Loudness Controller with reference to the input of the 1600's MB look-ahead limiter. (See *Loudness Control* on page 3-24.)

Loudness Controller Attack: See *Loudness Control* on page 3- 24.

Loudness Controller Bass Couple: See *Loudness Control* on page 3- 24.

Downmix Controls	
Name	**Range**
HF Limiter	–6.0…+6.0 dB
Drive Trim	–6.0…+6.0 dB
Loudness Threshold	–14.0…+10 dB, Off
Loudness Attack	0…100%
Loudness Bass Couple	0…12 dB, Off
BS1770 Safety Limiter	0…+6 dB, Off
Loudness Attack	0…100%
I/O CONTROLS:	
Downmix Center	–6.0…+3.0 dB
Downmix Surround	–6.0…+3.0 dB
Downmix Back	–6.0…+3.0 dB
Downmix 1 Target Loudness	–31…–5 LUFS
Downmix 2 Target Loudness	–31…–5 LUFS
Downmix Output Level	–6.0…0.0 dBFS
Downmix Proc Pre-emphasis	Flat, 50µs, 75 µs
Downmix Output Pre-Emphasis	Flat, Pre-emph

Table 3-14: Downmix Controls

<u>**BS.1770 Threshold**</u>: See *BS.1770 Safety Limiter* on page 3-26.

<u>**Downmix Center / Surround / Back**</u> (in I/O > Downmix) set the contribution of the center, surround, and back (7.1 only) channels to the downmix.

<u>**Drive Trim**</u> trims the drive level into the downmix HF limiter and loudness controllers over a range of ±6 dB. To permit the downmix loudness to track the surround loudness, the sum of the DRIVE TRIM and Surround MB LIMITER DRIVE controls sets the drive into the downmix HF limiter and loudness controllers.

Use the DRIVE TRIM control to make fine-tune the downmix loudness, which can be read from the downmix's loudness meters.

> This control is needed because the surround loudness meters and controllers operate on the power sum (RMS sum) of the surround channels, whereas the downmix is a voltage sum, where the mix of the channels is set by the I/O > DOWNMIX > DOWNMIX CENTER / DOWNMIX SURROUND / DOWNMIX BACK controls. This can cause the loudness of the downmix not to track the loudness of the surround.
>
> The best way to make sure that the loudness of the downmix never exceeds the DOWNMIX TARGET LOUDNESS is to activate the downmix loudness controllers and then adjust the DRIVE TRIM control so that a slight amount of loudness controller gain reduction always occurs for program material at normal levels. Normally, no more than 2 dB of DRIVE TRIM adjustment is needed to achieve this.

<u>**Downmix 1/2 Target Loudness**</u>: (in I/O > Downmix) These are gain controls between the output of the downmix loudness controller and the inputs of the output 1 and output 2 HF limiter / peak limiter chains. The controls scale the output of the loudness controller so that when the BS1770 Safety Limiter = 0 LU, the BS1770 loudness will be constrained to the Target Loudness value.

<u>**Downmix Output Level**</u> sets the maximum peak level produced by the processing. It does not adjust loudness. Do that with the Downmix #1 or #2 TARGET LOUDNESS control.

<u>**Downmix Proc Pre-emphasis**</u> determines if the input to the HF limiter and peak limiter is flat or if it has 50 µs or 75 µs pre-emphasis applied to it. Pre-emphasis is useful when driving analog channels like an analog TV transmitter.

<u>**Downmix HF Limiter**</u> Is a split band limiter that operates above 6 kHz to reduce pre-emphasis-induced overload when the downmix is being operated PRE-EMPHASIZED.

Test Modes

The Test Modes screen allows you to switch between OPERATE, BYPASS, and TONE. When you switch to BYPASS or TONE, OPTIMOD-PCn saves the preset you had on-air and will restore it when you switch back to OPERATE. Even if you had been editing a preset and did not yet save these changes as a User preset, you will not lose the edits you made.

Available Tone waveforms are SINE, SQUARE, and PINK NOISE.

The pink noise spectrum is accurate to ±0.05 dB, 20-20,000 Hz.

"Squarewaves" are available up to 1 kHz.

> Your Optimod is a band-limited system (like any digital system). Because of the Gibbs phenomenon, no band-limited system can produce true squarewaves without overshoot. Instead, your Optimod generates useful squarewave-like waveforms without overshoot by applying a waveform whose sample values periodically switch between +1 and –1 to a non-overshooting pulse-shaping lowpass filter. This eliminates overshoot at the cost of increasing the risetime of the "squarewave" edges.
>
> The main purpose of these waveforms is to test the transmission system following the Optimod to determine if the path introduces overshoot, tilt, or ringing into the "squarewave." This is particularly important if you are using an analog signal path after the Optimod because the –3 dB frequency of the path must be 0.15 Hz or lower to prevent 50 Hz squarewaves from overshooting more than 1% and increasing peak levels.
>
> To prevent aliasing and ensure that only odd-order harmonics are gener-

Setup: Test				
Parameter Labels	**Units**	**Default**	**Range (CCW to CW)**	**Step**
Mode	---	Operate	Operate, Bypass, Tone	---
Tone Waveform	---	---	Sine, Square, Pink Noise	---
Bypass Gain	dB	0.0	–18 … +25	1
Tone Frequency (Sine)	Hz	400	16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 9500, 16000, 12500, 13586.76, 15000, 20000	LOG
Tone Frequency (Square)	Hz	400	16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000	LOG
Tone Level	%	100	0 … 121	1
Lf Tone	---	Bypass	Off, On, Bypass	---
Rf Tone	---	Bypass	Off, On, Bypass	---
Ls Tone	---	Bypass	Off, On, Bypass	---
Rs Tone	---	Bypass	Off, On, Bypass	---
C Tone	---	Bypass	Off, On, Bypass	---
LFE Tone	---	Bypass	Off, On, Bypass	---
Lb Tone	---	Bypass	Off, On, Bypass	---
Rb Tone	---	Bypass	Off, On, Bypass	---

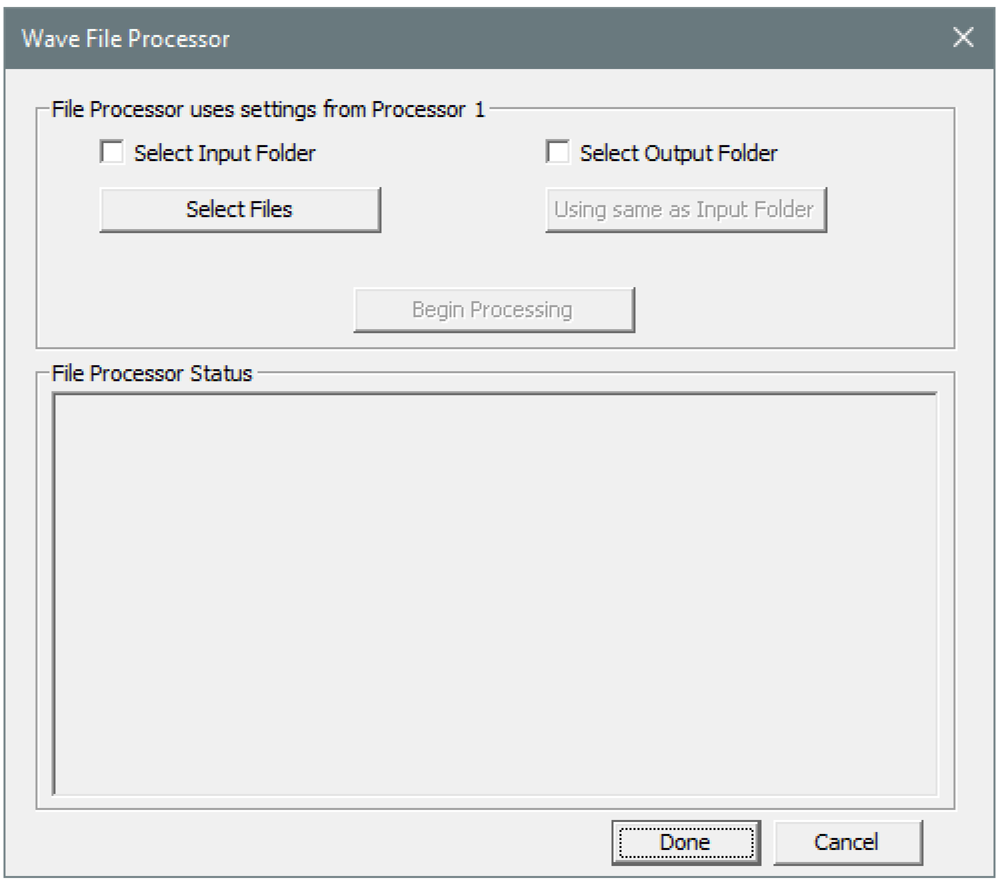
Table 3-15: Test Modes

ated, the source waveform is generated at 192 kHz sample rate and has an identical, integer number of samples in its “+1” and “–1” segments. Hence, output frequencies that are not submultiples of 96 kHz (like 315 Hz) will differ from their labeled values by a few percent. This causes no problems when testing transmission systems.

Table 3-15: Test Modes on page 3-80 shows the facilities available, which should be largely self-explanatory.

In TONE mode, any channel can be set to OFF (the tone is off). ON (the tone is on) or BYPASS (equivalent to placing that channel in BYPASS mode). When bench testing OPTIMOD-PCn, you can cause one hardware output to emit a tone while leaving the remaining channels in BYPASS. This eliminates the need for an external test oscillator.

Using Optimod-PCn for Offline File Processing



Optimod-PCn provides basic offline file processing functionality via the EDIT>PROCESS FILE menu item. Only 16-bit and 24-bit linear PCM stereo.wav files at 44.1 kHz, 48 kHz, 96 kHz, and 192 kHz sample rates are supported. To avoid compromising quality, MP3 is not supported.

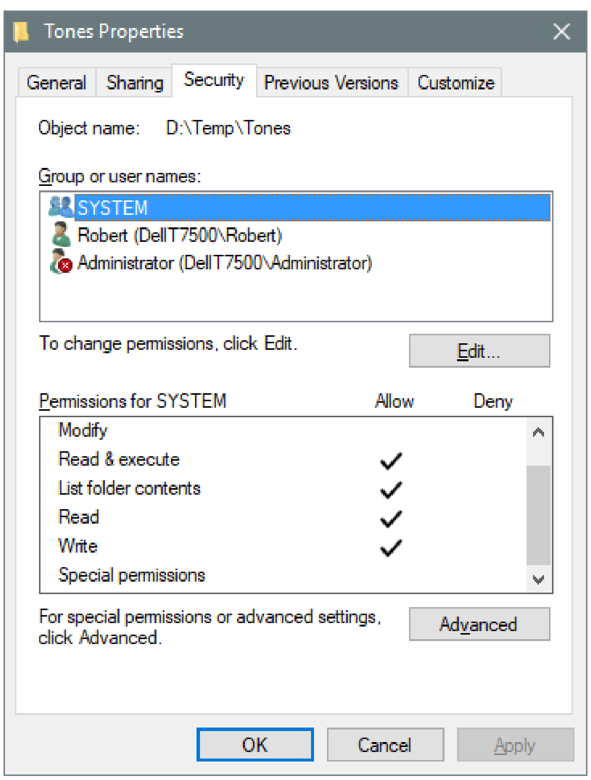
If the input file contains metadata such as INFO List, ID3v2, Cart Chunk (AES46-2002), and/or Broadcast WAVE Format (BWF) "bext" metadata, Optimod-PCn will copy this metadata to the output file.

File processing does not interrupt streaming on active Processors. To determine the number of Processors available for file processing, your Optimod subtracts the number of actively streaming processors (as shown in the Number of Processors field in Tools>Service Settings) from the total number of stereo plus surround processors available from your USB key or CodeMeter authorization (as shown in the Processor Configurations Available field in Tools>Service Settings) and dedicates the unused processors to file processing (up to a maximum of 16 processors). If you are using all of your available processors for streaming, your Optimod will create one "extra" processor and dedicate it to file processing.

Processing speed depends on the host computer's CPU and the number of Processors available for file processing. If the CPU is capable of glitch-free processing in your Optimod's live streaming mode, file processing will be somewhat faster than real-time. When processing all the files in a folder, your Optimod will open each file in an available processor and will process the files in parallel. For example, if you are processing a folder with four or more files and the Number of Processors is four, this will approximately quadruple the processing speed compared to having just one Processor.

You can choose to process a single file or all files in a specified folder. Output files have the same name as the input files except that `_out` is appended to the file name. You may specify separate Input and Output folders; if you do not specify an Output folder, the output files will be written to the Input folder.

The output file is the same size as the source file. Therefore, we recommend having at least 50 ms of silence at the beginning and end of the source file to accommodate both slight uncertainties in processing delay compensation and processes like the subharmonic synthesizer and multiband compressor that use filters whose transient response can slightly extend the active length of the output file compared to the input file.



Because the file processor runs as a Windows Service, *the folder containing the source and output files must have Administrator and System security access in Windows*. The Input folder must have Read permission and the Output folder must have Write permission. To verify and (if necessary) implement this, right-click the folder and choose PROPERTIES> SECURITY. The “Group or user names” box must include “Administrators” and “SYSTEM.” If not, click EDIT and then in the “Permissions for ‘Folder name’” box that appears, click ADD. After you have added “Administrators,” click it and check the READ and WRITE permissions in the “Permissions for Administrators” box. Do the same thing for SYSTEM.

File processing uses Processor 1’s current processing and system settings.

Your Optimod does not have a built-in player to preview file processing, so you must use an external player like Windows Media Player. Because file processing uses Processor #1’s settings, we suggest using Processor 1’s streaming functionality to preview and adjust the file processing. The rest of your available processors can be used for a combination of streaming and file encoding

A) To preview, use the CONNECT menu to connect to Processor #1.

B) Open the file in the external player and play it out while listening to the Optimod’s output. (For example, you could connect the player to the Optimod’s Processor 1 input via Virtual Audio Cable.) While listening, choose and/or customize a preset to taste.

> Note that if you are preparing files for a playout system to feed a stream having no online audio processing, you must know the target loudness of the stream and must set the Optimod’s TARGET LOUDNESS (in the 2.0 LESS-MORE tab) to match. Doing this will calibrate the Optimod’s loudness meters so that “0” corresponds to the target loudness. If you are customizing a preset, you may use these meters to guide your adjustments so that the loudness of the processed files will be correct. See *About Target Loudness and ITU-R BS.1770* starting on page 1-16.

> Also note that file processing cannot be aware of boundaries and transitions between program elements contained in multiple files. For example, if you mix two loudness-normalized files, the loudness of the mix can be 3 (or more) dB higher than target loudness. For this reason, we believe that for streaming applications, online, live Optimod-PCn processing is likely to provide the best possible listening experience.

C) Once you have tuned the processing to your liking, choose the FILE>PROCESS FILE menu item. This will temporarily disconnect the GUI from Processor 1.

D) Using the SELECT INPUT FOLDER and SELECT OUTPUT FOLDER checkboxes and SELECT FILES or SELECT FOLDER button, choose the file or folder you wish to process and the destination folder.

> If you check SELECT INPUT FOLDER, your Optimod will process all the source files in the folder that you choose after you click SELECT FOLDER. If you do not check SELECT INPUT FOLDER, your Optimod will process just the file you choose after you click SELECT INPUT FILE.

E) Click BEGIN PROCESSING. Any streaming that is occurring on other Processors will continue, and the File Process Status Window will display the processing’s progress.

Until the processing is complete and you have clicked the DONE button (or until you click the CANCEL button), you will not be able to reconnect the GUI to a

Processor. If you were connected to Processor 1 prior to invoking the FILE>PROCESS FILE menu item, clicking the DONE or CANCEL buttons will re-connect the GUI to the Processor automatically.

> If the CANNOT OPEN FILE error message appears, correct the security permissions for the input and output folders per the instructions on page 3-82.

Figure 3-17 shows the typical file processing speed for a 6th-generation i7 CPU as a function of CPU clock speed and the limiting algorithm in use: no MX, MX SOFT, and MX HARD. The measurements for this graph were made on a Dell Precision 3620 with a solid-state drive, 16 GB of RAM, and an Intel i7 7700K 4.2 GHz (4.37 GHz measured) CPU running Windows 10 Pro. The "Processing Time" as a function of CPU clock speed was based on the 4.37 GHz measurements with the assumption that processing speed is linearly proportional to clock frequency.

Using the Graph: To estimate the time required to process a given file, on the graph find the "processing time" based on your CPU speed and limiter algorithm; then divide the real-time speed of your file by the speed-up factor. For example, if your file is 5 minutes long, your CPU is clocked at 4.0 GHz, and the MX limiter is off in the active preset, the curve shows a speed-up of about 6x, so the file will take approximately 5/6 minutes (50 seconds) to process.

Note that there is an initialization time of 5-10 seconds before the file processing starts, so counting initialization, shorter files will produce less speed-up than longer files.

Using the API for File Processing

Your Optimod's API provides commands to process files. This allows you to automate

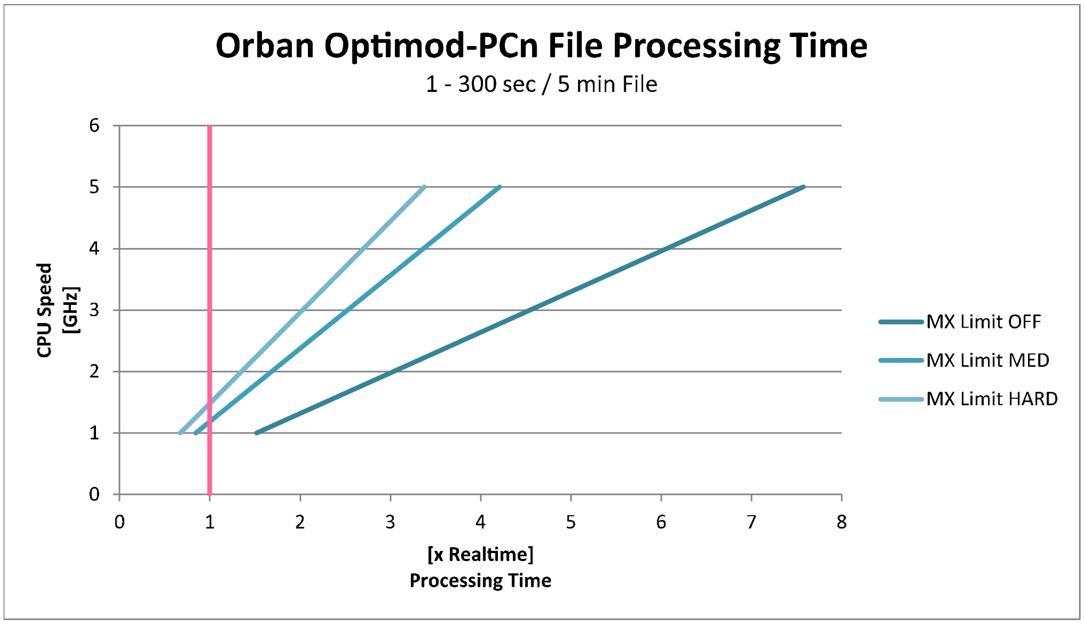


Figure 3-17: Normalized File Processing Time as a Function of CPU Speed

file processing by writing custom software to generate the required commands. See *OPTIMOD 1600PCn Control API* starting on page 5-1.

These commands are case-sensitive.

Start by specifying the input file or folder.

For processing a single file:

```
fpinput c:\music_source\sourcefile.wav
```

c:\music_source\sourcefile.wav is the full path of the source file.

returns

```
FP_INPUT:source source file found [or] source file not found
```

For processing a folder:

```
fpinput c:\music_source\
```

c:\music_source\ is the source folder.

returns

```
FP_INPUT:source source folder found [or] source folder not found
```

After you have specified the input file or folder via the `fpinput` command, specify the output folder. As soon as you do this successfully, file processing will start:

```
fpoutput c:\music_destination\
```

returns

```
FP_OUTPUT:destination folder found. File processing started. [or] destination
folder not found
```

After a single file is processed (or an error occurs), the API returns

```
FP_STATUS:file processed successfully [or] Could not open destination folder
for writing. Check folder security permissions.
```

After a folder is processed (or an error occurs), the API returns:

```
FP_STATUS:folder processed successfully [or] Could not open destination folder
for writing. Check folder security permissions.
```

Production and Mastering Applications

Using its offline wav file processing capabilities (see *Using Optimod-PCn for Offline File Processing* on page 3-81), OPTIMOD-PCn can be a useful tool for mastering and production applications in the professional audio industry, such as preparation of equalized, level-controlled, peak limited CD masters. We have frequently used OPTIMOD-PCn in this context, achieving excellent results.

Because of their broadcast origins, most of the 1600's presets provide more processing than would ordinarily be required for mastering. In addition, we would expect that the mastering engineer would want to tweak a preset carefully to complement the program material being mastered. The 1600 provides important tools to allow a mastering engineer to fine-tune the processing to complement the program material:

- Four bands of parametric equalization with low-noise filter structures and curves modeled after classic second-order analog bell-shaped EQ.
- Sweepable lowpass and highpass filters with selectable 6, 12, 18 or 24 dB/octave slopes.
- A powerful, low-noise parametric bass shelving equalizer with sweepable frequency and a choice of 6, 12, or 18 dB/octave slopes.
- Five-band compressor/limiter with selectable allpass or phase-linear crossovers and powerful controls, including attack time, release time, threshold, knee, and ratio for each band. These compressor/limiters also offer user-adjustable inter-band coupling, allowing the user to operate them anywhere from quasi-wideband to fully independent. The compressor can be operated in either inline or parallel modes. In parallel mode, it can bring up the level of quiet material very unobtrusively while preserving the dynamics of louder material.
- An extremely powerful peak limiter with advanced, proprietary distortion reduction algorithms.

You cannot create a user preset "from scratch"; you must create it by modifying an existing preset, factory or user. Each preset has an "easy adjustment" facility called LESS-MORE, which is a one-knob provision for turning the amount of processing up or down.

Systematically, the following is a good method for creating mastering presets. It assumes that you have already set the processor mixer controls to achieve normal drive levels.

Note that by drawing a marquee around several controls by clicking and dragging with your computer's mouse, you can temporarily couple them so you can adjust several bands' controls (for example, the LIMITER ATTACK controls) simultaneously. The coupling will remain active until you click outside the area that the marquee encloses.

A) Set the 100% OUTPUT LEVEL control in the I/O Mixer to match your application. If your master is intended for a distribution channel using lossy compression, −1.5 dBFS is reasonable. For a lossless or PCM channel, −0.2 dB should be adequate.

    If your mastered audio is intended for transmission via a lossy codec like AAC, MP3 or WMA, be aware that the codec's decoder may overshoot and cause audible clipping distortion. See *Setting Output/Modulation Levels* on page 1-38. If the 1600's output is applied directly to a lossy codec, decrease the setting of the 1600's output level control to allow the necessary headroom. If the 1600's output is applied to a linear PCM storage medium (like a CD), it is better to use the entire dynamic range of the linear medium by setting the 1600's output level close to 0 dBFS. Then compensate for codec overshoots by reducing gain when you transcode from the linear PCM recording to the codec.

B) Recall a mastering-oriented preset. See the discussion of the PEAK LIMIT MX, SOFT KNEE, SOFT KNEE MX, ROCK MASTERING MX, and ROCK MASTERING MXH presets in *Protection, AGC and Mastering* Presets starting on page 3-34. These presets are good starting points, but we would expect a mastering engineer to adjust them to taste.

    If you want to use parallel compression mode, start with CLASSICAL PARALLEL MX and turn off the AGC. (You can do this in the preset's AGC tab or by setting EXTERNAL AGC TO YES in I/O > UTILITY.) See *Parallel Compression* on page 3-21.

    Note that you can use the PASS-THROUGH SWITCH (in the LESS-MORE tab) to compare your work with the original audio.

C) In the LESS-MORE tab, set the TARGET LOUDNESS according to the requirements of your deliverable. This control scales OPTIMOD-PCn's LOUDNESS LEVEL meters so that "0" on these meters corresponds to the active TARGET LOUDNESS.

    The BS.1770 meter (the rightmost LOUDNESS LEVEL meter) shows BS.1770-2 (or higher) Integrated loudness as a varying-length bar and Short-Term loudness (3-second integration time) as a single white segment. The Integrated loudness is gated according to BS.1770-2 and is computed using a 10-second rolling window of integration.

    If the medium uses the Dolby Digital codec, set OPTIMOD-PCn's TARGET LOUDNESS REFERENCE value to match the dialnorm value to be authored into the medium's Dolby Digital metadata. (Refer to step 4.C) on page 2-18.)

D) The mastering-oriented presets have the AGC turned OFF. If you need a very large amount of compression for an application like processing material intended for in-flight entertainment systems, you can either edit the preset to turn the AGC on or start with a non-mastering preset.

    You can turn the AGC off globally for all presets, which is convenient if you don't expect to use it in the future. See step (2.C) on page 2-16.

E) Unless you will be using a large amount of compression for special applications, set the MB GATE THRESHOLD to OFF.

    In the mastering-oriented presets, this is already done.

F) Adjust the MULTIBAND DRIVE control to achieve the desired amount of multiband gain reduction. Except for PEAK LIMITER MX, the mastering-oriented presets will adjust this control as LESS-MORE is adjusted.

G) Adjust the MULTIBAND RELEASE control to achieve the desired compression density.

You can use the DELTA RELEASE controls to fine-tune the release time of each band independently.

The release characteristic is always "automatic" (i.e., with program-dependent time constants) and the MULTIBAND RELEASE control simply scales this process. This, combined with multiband operation, makes the compression remarkably resistant to the usual compressor pumping and squashing.

H) Adjust the ATTACK TIME controls on the individual compressors to trade off overshoot control against transient punch.

I) Adjust the RATIO and KNEE controls in each band to taste.

The RATIO control sets the compression ratio at the threshold of compression. To achieve a classic soft knee characteristic, set the RATIO to 1:1 and adjust the softness of the knee with the KNEE control.

The KNEE control sets the gain reduction in dB at which the compression ratio reaches ∞:1. (See Figure 3-16 on page 3-71 and associated text for a description of the RATIO, KNEE, and BREAKPOINT controls.)

J) After you adjust the RATIO and KNEE controls, adjust the THRESHOLD controls in the individual bands to achieve the desired amount of gain reduction.

The KNEE control automatically and invisibly changes a given band's internal compression threshold to keep the compressor's output level constant whenever the drive level is high enough to move the gain reduction into the ∞:1 range. This means that the internal threshold decreases with softer knee settings (higher settings of the KNEE control). However, the indicated threshold the 1600's user interface does not change. This behavior ensures that the THRESHOLD control alone determines the maximum output level of the compressor, regardless the KNEE control's setting.

K) Adjust the LIMITER ATTACK TIME controls to taste.

These controls allow you to set the limiter attack anywhere from 0% to 100% of normal in the Five-Band compressors, each of whose gain reduction has a fast-release (limiter) and slow-release (compressor) component. Because the limiter and compressor characteristics interact, you will usually get best audible results when you set these controls in the range of 50% to 100%. Below 50%, you will usually hear pumping because the compressor function is trying to create some of the gain reduction that the faster limiting function would have otherwise achieved. If you hear pumping in a band and you still wish to adjust the limiter attack to a low setting, you can sometimes ameliorate or eliminate the pumping by slowing down the compressor attack time in that band.

Of course, sometimes artistic pumping is desired for certain styles of music and/or recording. The LIMITER ATTACK TIME controls can help achieve this sound.

If you are using low compression ratios and small amounts of gain reduction, setting the LIMITER ATTACK TIME controls as low as 50% can often increase punch. This is the typical setting in the mastering-oriented presets.

L) Adjust the TRANSIENT ENHANCE control to taste.

This control allows you to insert an audio delay of up to 10 ms in the sidechain of the five-band compressor. Delaying the gain control signal, allows attack transients to pass through the multiband compressor uncompressed, which can increase punch. There is a tradeoff between this control and the amount of gain reduction in the peak limiter, which will have to eliminate attack transients exceeding the peak limiter's threshold.

M) Adjust equalization as necessary.

As discussed above, there is a versatile program equalizer available between the AGC and multiband compressor. There is also a five-band mix control (functioning as a graphic equalizer) after the five-band compressor. Note that any fixed equalization will be partially "undone" by the dynamic re-equalization effect of the five-band compression.

You can use the individual band compression threshold controls and the various BAND COUPLING controls to affect the amount of automatic re-equalization performed by the five-band compression and the spectral balance it produces. As you set the BAND COUPLING controls closer to 100%, they permit progressively less dynamic program-adaptive equalization. If you feel that the dynamic re-equalization is not producing enough brightness when the program material lacks high frequencies, you should turn the BAND 3>4 and BAND 4>5 COUPLING closer to 0%. Similarly, if weak bass is not sufficiently boosted, turn the BAND 2>1 COUPLING closer to 0%.

N) Set the amount of peak limiting with the MB FINAL LIMIT DRIVE control.

In general, the less peak limiting you use, the better sounding the result will be. However, if your client demands a "loud" result, the 1600's MX peak limiter is a powerful tool for achieving this with minimum distortion or other side effects. Moreover, when driven hard, the MX limiter can add useful and attractive spectral components that can increase the perceived detail in the master.

See *Peak Limiter* on page 3-23 for a further discussion of the peak limiting technology available in OPTIMOD-PCn.

O) Adjust the BASS CLIP THRESHOLD, BASS PRE-LIMITING, and BASS LIMITING controls to complement the amount of final limiting. The values in the master-oriented factory presets are usually satisfactory. The effect of these controls will depend on the amount of peak limiter gain reduction, as set by the MB FINAL LIMIT DRIVE control. If you drive the MX limiter too hard, it will reduce bass punch in an attempt to control IM distortion and to preserve essential midrange information.

For an explanation of the MX bass limiting controls, see *Peak Limiter* starting on page 3-73.

P) Save your preset using File/Save Preset.

Once you have created one "mastering" preset, you can edit it to create others and save them under different names.

Q) Process your audio file, following the instructions in *Using Optimod-PCn for Offline File Processing* on page 3-81.

Creating Custom "Factory" Presets

You can create custom "factory" presets that support LESS-MORE functionality. These presets will behave like the factory presets.

OPTIMOD-PCn's software has the ability to interpolate all 19 available LESS-More increments (1.0 through 10.0) from two or more LESS-MORE "anchor" preset files. Additionally, you must create a "factory preset" file that corresponds to the default value of LESS-MORE for your preset. The instructions below show you how to create these files from user preset files that you have saved.

In the examples below, the specified file locations are based on OPTIMOD-PCn's default locations. Using Windows Explorer, check the factory preset file locations on your computer to find out where these files are actually located. You must place your new "factory preset" files in analogous locations.

To create a custom "factory" preset:

1. **Make a working directory for your preset.**

   When you have finished making it the preset, you will copy it into the OPTIMOD-PCn folder in the Windows `C:\Program Files (x86)\Orban\Optimod-PCn` 1600 directory.

2. **Choose a name for the preset.**

   You cannot use the name of an existing factory preset without damaging it. We will use "My Preset" as the preset name in the examples below.

3. **Create a folder for your presets.**

   Create a folder named "My Preset" within your working directory.

4. **Choose the default LESS-MORE value for your preset.**

5. **Using 1600PC, create user presets for each LESS-MORE anchor point.**

   See *Customizing Processing Presets* starting on page 1-29.

   Because LESS-MORE works only on the dynamics processing, you must choose a set of equalizer settings for your new preset that is the same for all anchor presets. *Table 3-8: Equalizer Controls* on page 3-56 shows the controls that must be set identically in all anchor presets.

   You must create at least two anchor presets, one for LESS-MORE = 1.0 and one for LESS-MORE = 10.0. Additionally, if the default LESS-MORE value you chose in step 4 is neither 1.0 nor 10.0, you must create a third anchor preset for the default LESS-MORE value.

You may add as many additional anchor presets as you wish to fine-tune the behavior of LESS-MORE.

From the OPTIMOD-PCn Control Application, save each preset as a user preset in plain text (unencrypted) form, naming it `My Preset LMxxx`, where `xxx` stands for the LESS-MORE setting that corresponds to that preset. For example, for LESS-MORE = 1.0, `xxx` = 010.

6. **Copy the user presets you made.**

   Copy these presets in the `\[working directory]\My Preset\` folder that you made in step 3.

7. **Edit the file extensions.**

   In Windows, change the extension of all of your presets' file names to `orb1600f`

8. **Edit the text within the preset files.**

   A) In turn, open each of your preset files in a text editor like Notepad or WordPad. The editor you use must open and save files in plain text format.

   B) Edit the second line of the file as follows:

   ```
   Preset Name=<MY PRESET LMxxx> size=[yyy]
   ```

   where `LMxxx` agrees with the file name.

   Do not change the number after `size`.

   A) Edit the third line of the file as follows:

   ```
   Factory Preset Name=<MY PRESET>
   ```

   B) Edit the fifth line of the file as follows:

   ```
   Preset File Name= <MY PRESET LMxxx.orb1600f>
   ```

   where `MY PRESET LMxxx.orb1600f` is the same as the file name.

   C) At the end of the file, find the "LESS-MORE" line and edit it as follows:

   ```
   C:<LESS MORE>Cent:x;D:y;
   ```

   Replace the "x" and the "y" with the LESS-MORE data corresponding to the LESS-MORE value that you chose for your new preset. See **Table 3-16**.

   D) Save your edited file as a plain text file. (This should happen automatically if you edited the file using a text editor, not a word processor.)

   E) Repeat steps (A) through (D) for each anchor file you created.

LESS-MORE	x	y
1.0	100	0
1.5	150	1
2.0	200	2
2.5	250	3
3.0	300	4
3.5	350	5
4.0	400	6
4.5	450	7
5.0	500	8
5.5	550	9
6.0	600	10
6.5	650	11
7.0	700	12
7.5	750	13
8.0	800	14
8.5	850	15
9.0	900	16
9.5	950	17
10.0	1000	18

Table 3-16: LESS-MORE Reference

9. Create the factory preset file.

A) Copy the LESS-MORE file corresponding to your preset's default LESS-MORE setting, placing the copied file in

`\[working directory]`

B) Edit the file name of the file you copied to remove `LMxxx`, including the space before the "L." The example file name is

`MY PRESET.orb1600f`

C) Open `MY PRESET.orb1600f` in a text editor.

D) Edit the second line of the file as follows:

`Preset Name<MY PRESET> size=126`

Do not change the number after `size`.

E) Edit the fifth line of the file as follows:

`Preset File Name= <MY PRESET.orb1600f>`

F) In Windows, give all of the files you just created the "read-only" attribute.

10. Copy your new preset to the OPTIMOD-PCn program file directory.

Copy both the `MY PRESET` folder and the `MY PRESET.orb1600f` file to `C:\Program Files (x86)\Orban\Optimod-PCn 1600\presets`.

Because this modifies files and folders within the Windows program file directories, you need Administrator privileges to do this.

Your new "factory preset" is now ready for use. When you recall a preset via the OPTIMOD-PCn Control Application, the new preset should appear in the OPEN PRESET window. After you recall it, LESS-MORE should be available.

The most common mistake is to put the incorrect LESS-MORE value in the My Preset.orb1600f file or its corresponding LESS-MORE file. This will cause LESS-MORE not to work.

Back up your new "factory" preset (i.e., the factory preset file and less-more anchor files) elsewhere. If you install a new version of OPTIMOD-PCn, it will overwrite your "factory" preset and you must restore it from a backup.

Appendix A: Using the ITU BS.1770 and CBS Loudness Meters to Measure Loudness Controller Performance

[**Note:** This Appendix is a reprint of a stand-alone white paper. For this reason, it contains some explanatory material regarding the CBS and BS.1770 meters that is also found elsewhere in this manual.]

ITU-R BS.1770

In 2009, the ATSC released a Recommended Practice: Techniques for Establishing and Maintaining Audio Loudness for Digital Television (A/85:2009). This was later up-dated as A/85:2011. A/85 specifies use of a long-term loudness meter based on the ITU BS.1770 algorithm for measuring the loudness of DTV broadcasts.

In December 2011, the FCC adopted rules implementing the CALM Act, which, by law, forbids commercials from being louder than non-commercial program material. The new FCC rules incorporated ATSC A/85 (and, by implication, the BS.1770 meter) as an objective means of verifying that the rule was being obeyed.

Because loudness measurement per BS.1770 uniformly integrates all program material, quiet passages tend to lower the measured value. To prevent this, the ITU added gating to the BS.1770 standard, which was revised as BS.1770-2 in March 2011. The gating causes the meter to ignore silence and to integrate only program material whose loudness falls within a floating window extending from the loudest sounds within the specified integration period to sounds that are 10 dB quieter than the loudest sounds. This is because humans tend to assess loudness based on the louder sounds in a given program. (The revision current as of this writing, BS.1770-4, added the ability to monitor systems with more than 5.1 channels.)

The ATSC A/85 2011, ITU-R BS.1770-4, and EBU R 128 documents are available as free downloads. A search engine like Google can locate them easily.

CBS Loudness Meter

For many years, Orban has used the Jones & Torick loudness controller and loudness measuring technology[4] in its products for loudness control of sound for picture. Developed after 15 years of psychoacoustic research at CBS Laboratories, the CBS loudness controller accurately estimates the amount of perceived loudness in a given piece of program material. If the loudness exceeds a preset threshold, the controller automatically reduces it to that threshold. The CBS algorithm has proven its effectiveness by processing millions of hours of on-air programming and greatly reducing viewer complaints caused by loud commercials.

[4] Jones, Bronwyn L.; Torick, Emil L., "A New Loudness Indicator for Use in Broadcasting," J. SMPTE September 1981, pp. 772-777.

Since first licensing the CBS algorithm and using it in its Optimod-TV 8182 in the early 1980s, Orban has continually refined and developed this technology. In the last 30+ years, audio processors from Orban and CRL using the CBS loudness controller have processed millions of hours of on-air television programming — an unsurpassed track record that no other subjective loudness controller technology can claim.

Comparing the Meters

Because the ATSC recommends the BS.1770 algorithm, many broadcast and cable engineers facing the problem of controlling broadcast loudness have wondered how the CBS and BS.1770 technologies compare. An earlier version of this Orban white paper compared the CBS and BS.1770-1 (non-gated) meter. The paper you are now reading was revised in March 2012 to incorporate results from tests using the BS.1770-2 algorithm and EBU – TECH 3342 "Loudness Range" algorithm. The new measurements were performed using Version 2 of the Orban Loudness Meter[5]. This revision compares the CBS and BS.1770-2 meters because we expect that the ATSC will eventually update A/85 to specify BS.1770-2, which will more closely harmonize A/85 with its European counterpart, EBU R128.

A/85 and R128 differ significantly in their philosophy and recommendations. Probably most important difference is that A/85 asserts that the loudness of a so-called "anchor element" (which is typically dialog except in programs emphasizing music, like live concert recordings) is most important, while R128 asserts that the integrated loudness of the entire program is most important[6] and therefore, program loudness should be normalized based on an integrated BS.1770-2 measurement. The philosophy behind A/85 is similar to that of Dolby Laboratories, which for many years has asserted that dialog anchors most film and television programs and that listeners set their volume controls to make dialog comfortably intelligible[7]. (We agree more with A/85 than with R128).

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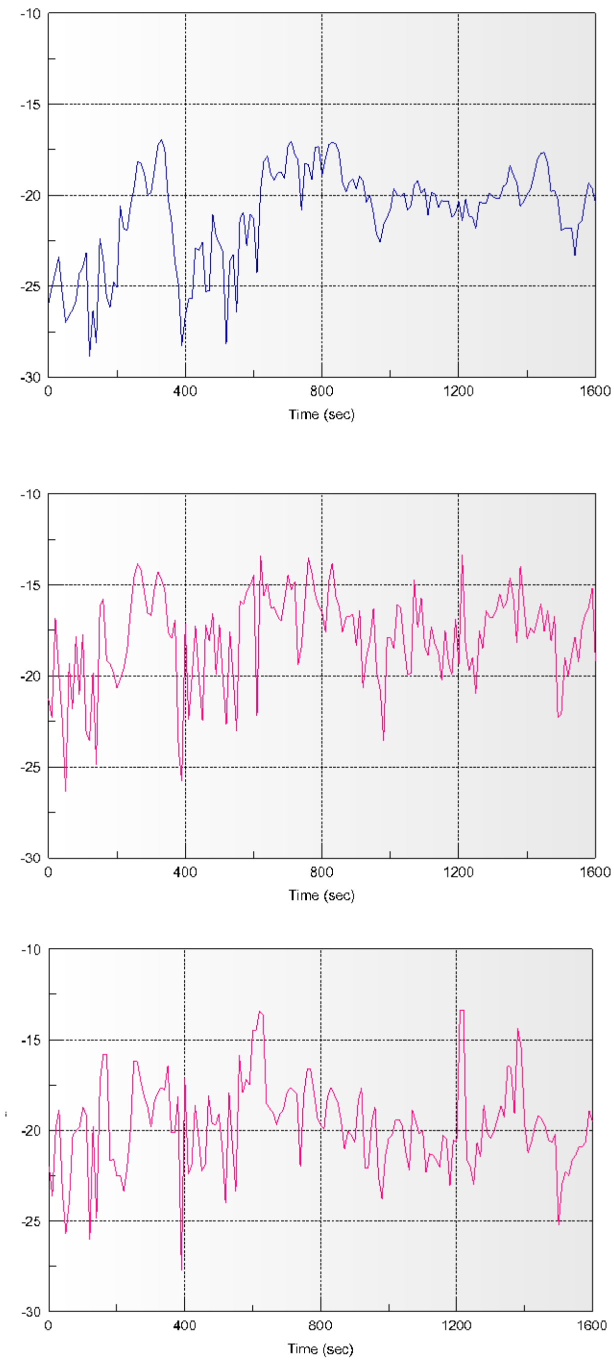
[5] This software is available for free download at http://orban.com/meter/.

[6] EBU – TECH 3343, "Practical guidelines for Production and Implementation in accordance with EBU R128," version 1 (February 2011), p. 29

[7] Riedmiller, J., Lyman, S., Robinson, C., "Intelligent program loudness measurement and control: what satisfies listeners?" AES Convention Paper 5900, 115th Convention (October 2003)



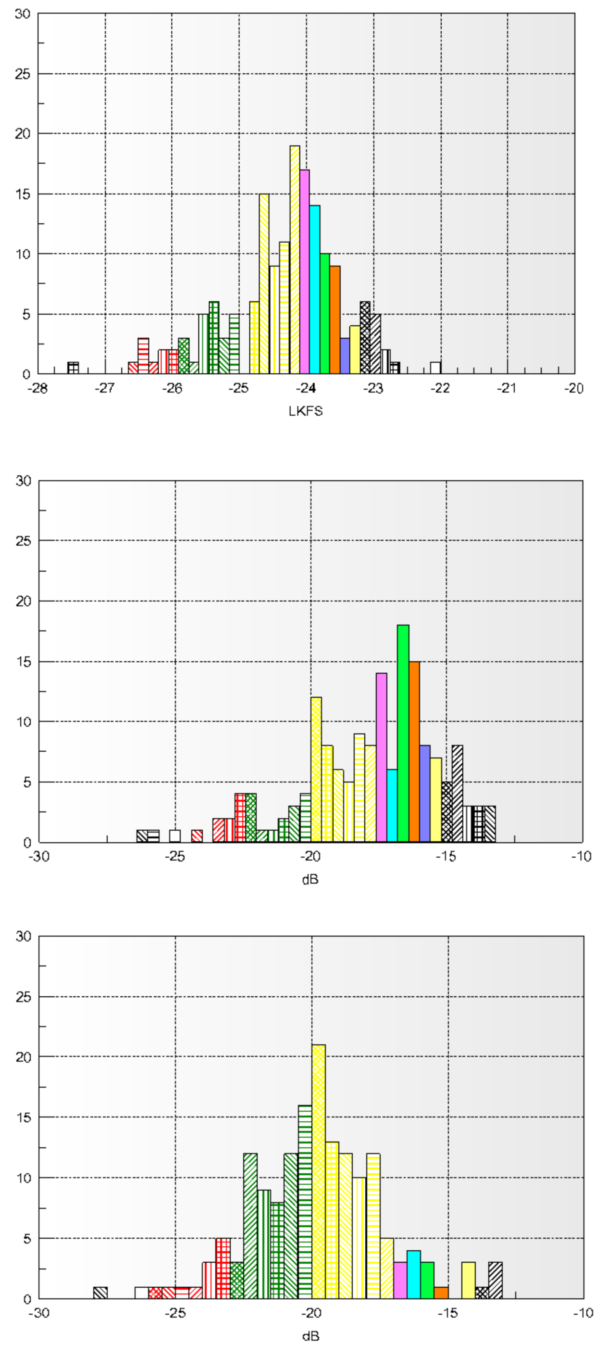


Figure 3-18: Unprocessed Input—
Peak Output of the BS.1770 and CBS Loudness Meters in each 10-second Interval as a Function of Time

Figure 3-19: Unprocessed Input—
Histograms sorting loudness measurements into 0.25 dB bins.

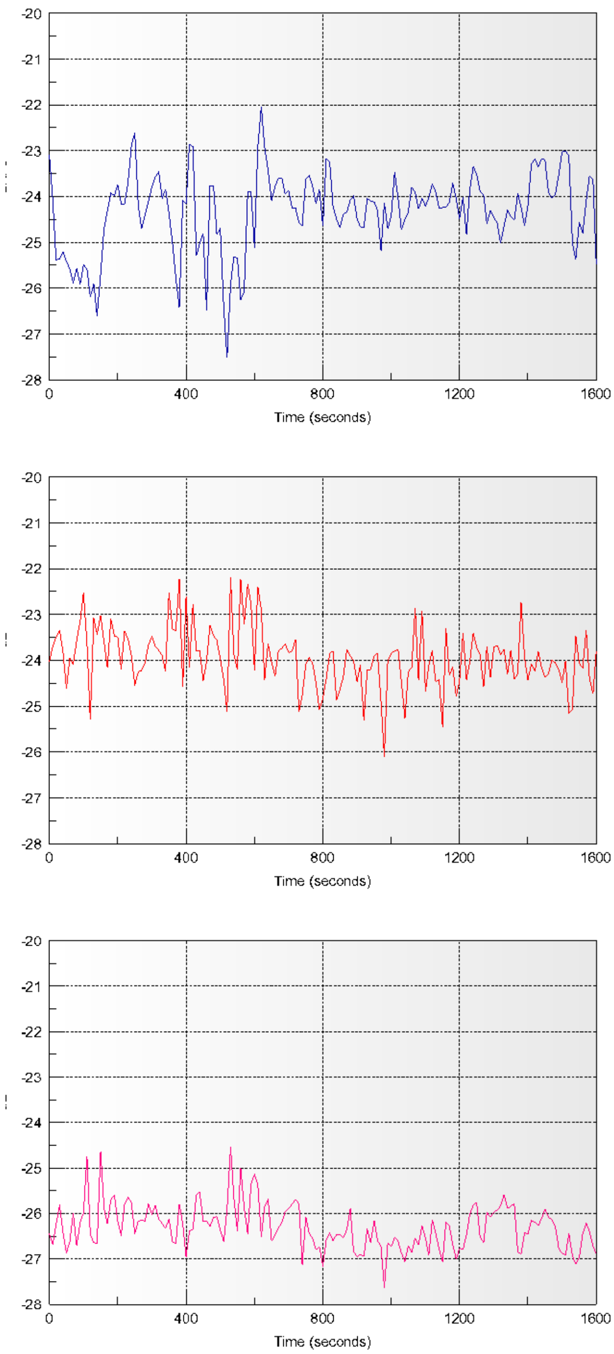


Figure 3-20: Loudness-Controlled Audio—
Peak Output of the CBS and BS.1770 Loudness Meters in each 10-second Interval vs. Time

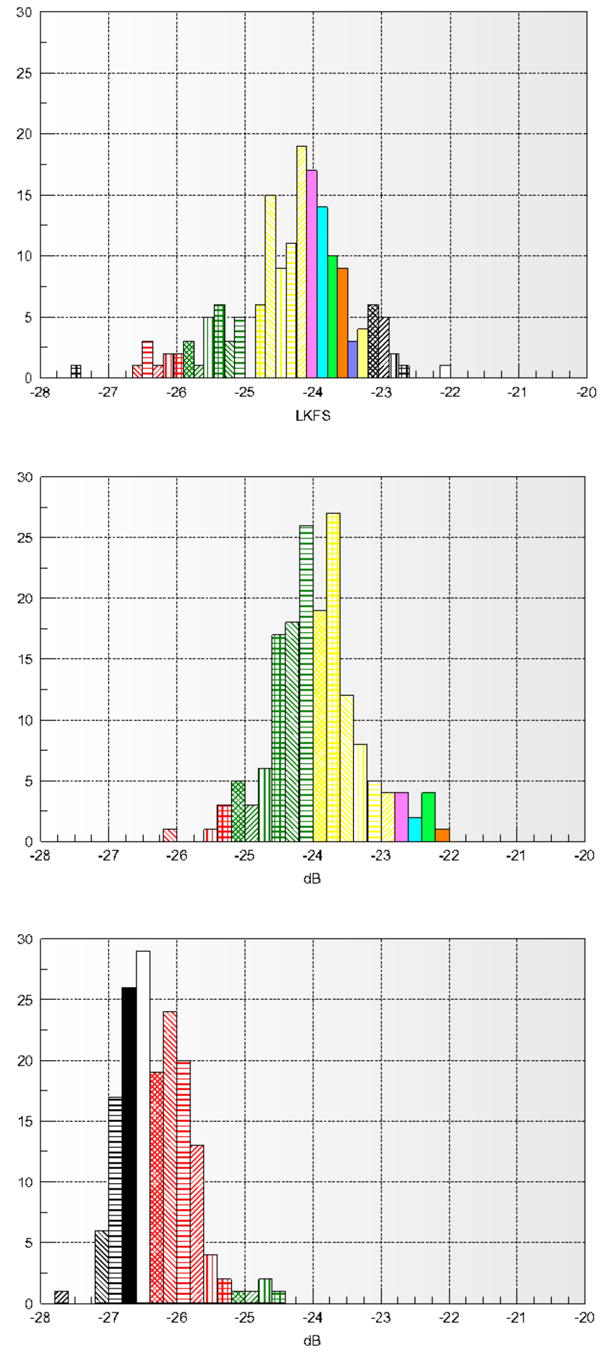


Figure 3-21: Loudness-Controlled Audio—
Histograms sorting loudness measurements into 0.25 dB bins

Test Setup

A stereo recording of approximately 30 minutes of unprocessed audio from the output of the master control of a San Francisco network station was applied to the 2.0 processing chain of an Optimod-Surround processor, set for normal operation using its TV 5B GEN PURPOSE preset. The digital output of the processor was applied to the digital input of an Orban 1101e soundcard, which was adjusted to pass the audio without further processing and to apply it to a software-based loudness meter that simultaneously computes the BS.1770-2 Integrated loudness and CBS loudness. The first 750-second segment of the program material was a daytime drama with commercial and promotional breaks, while the remainder was local news, also with commercial and promotional breaks.

The BS.1770-2 meter was adjusted to produce a 10-second integration window in which, per the BS.1770 standard, all data are equally weighted. The CBS Loudness Gain control was set to −3.12 dB. Data were logged every 10 seconds and included the maximum meter indication produced by both the BS.1770 and CBS meters in each 10-second interval. This produced 165 data points, which were imported into a scientific plotting application[8].

Orban's experimental long-term loudness measurement, based on the CBS meter and first published in 2008, was also included in the measurements and is shown in the bottommost charts. This algorithm attempts to mimic a skilled operator's mental integration of the peak swings of a meter with "VU-like" dynamics. The operator will concentrate most on the highest indications but will tend to ignore a single high peak that is atypical of the others. This algorithm can be seen to share certain characteristics with the floating gate first introduced in EBU R128 and later adopted in BS.1770-2.

The Orban algorithm displays the average of the peak indications of the meter over a user-determined period: 10 seconds for these measurements. The average is performed before dB conversion. All peak indications within the period are weighted equally with the following exceptions:

- If the maximum peak in the window is more than 3 dB higher than the second highest peak, it is discarded.
- All peaks more than 6 dB below the maximum (or second-to-maximum, if the maximum peak was discarded) are discarded.

Because the CBS long-term measurement discards a single peak if it is more than 3 dB higher than the second highest peak, the CBS long-term measurement tends to be biased about 3 dB lower than a measurement that shows the maximum peak indication of the CBS meter in a 10-second period. This depends on whether or not the loudness applied to the meter's input is well controlled. This bias can be seen in the figures in the paper. Because the Orban meter allows control of the level applied to

[8] PSI Plot: http://www.polysoftware.com/plot.htm

the CBS algorithm via the "CBS Gain" control, setting it 3 dB higher could better match the CBS long-term measurement to the BS.1770-2 Integrated measurement at the expense of moving the "maximum peak loudness" indication 3 dB higher.

Results

Unprocessed Audio Input: To provide a baseline for discussion of the loudness-controlled results, we measured the unprocessed audio that was applied to the Orban processor's input. Figure 3-18 and Figure 3-19 on page 3-95 show the loudness of the unprocessed audio both as a function of time and as a histogram. The histogram sorts the meter outputs into 0.25 dB or 0.25 Lk[9]-wide slices and shows the number of measurements that fit into each of these slices. The histogram thus portrays loudness consistency — when the histogram is clustered tightly within a few bins, the loudness is more consistent than it is when the histogram is spread out into a larger number of bins.

With all meters, the histogram of the unprocessed audio shows a wide spread. This is consistent with the EBU Loudness Range measurement for the entire clip, which was 16.5 Lk, while the LRA for the daytime drama alone was 19.2 Lk (including commercials). The BS.1770-2 Integrated loudness was −20 LUFS, integrated over the entire measurement period, although the inconsistencies between the loudness of program material and commercials are large enough to make this 30-minute measurement essentially meaningless.

In general, the loudest parts of the unprocessed audio are commercials and promos, both network and local. These are anywhere from 5 to 10 dB (or Lk) louder than the rest of the program material. This inconsistency was not a problem because the station in question was using an Orban automatic loudness controller on-air, which smoothed out loudness differences before its input.

While the general shapes of the CBS and BS.1770 loudness vs. time curves are similar, there were some significant differences. For example at approximately 1250 seconds, the CBS measurement shows a sharp loudness spike that was caused by a network news report that was equalized to emphasize frequencies around 2 to 3 kHz, where the ear is most sensitive. The BS.1770-2 measurement did not indicate this as being louder than the surrounding program material although to our ears, it clearly was.

Loudness-Controlled Audio: Figure 3-20 and Figure 3-21 on page 3-96 show the results after automatic loudness control. (To present the data with optimum graphic resolution, we made the loudness scales of Figure 3-20 and Figure 3-21 narrower than the scales in Figure 3-18 and Figure 3-19.)

Both the loudness vs. time graphs and the histograms show the Orban processing controls loudness well, although the details of the meters' indications are different.

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[9] Unfortunately, two terms for the same loudness units have been used in different standards documents. Lk and LkFS (used in ATSC A/85) are the same as LU and LUFS (used in EBU R128 and BS.1770).

Both the BS.1770 and CBS measurements indicate that most of the data points are in a ±1 dB/LU window.

The peak CBS readings fit within a ±2 dB window. The BS.1770 readings also fit within a ±2 Lk window except for four short intervals, which appear as low-probability outliers in the left side of the histogram. These intervals correspond to dialog without background music and in the author's opinion illustrate a weakness in BS.1770-2: based on our extensive listening tests, we have concluded that the meter does not effectively lock onto the A/85 "anchor element" (almost entirely dialog in the test material used to prepare this paper) and instead indicates that loudness increases when dialog level is held constant while underscoring or effects are added to the mix.[10]

Problems with Low Peak-to-RMS Ratio Material

In the subjective testing to validate the BS.1770 meter, there were outliers as large as 6 dB (i.e., the meter disagreed with human subjective perception by as much as 6 dB[11].) The subjective testing to validate the CBS meter found outliers up to 3 dB, although fewer items were used in this testing.

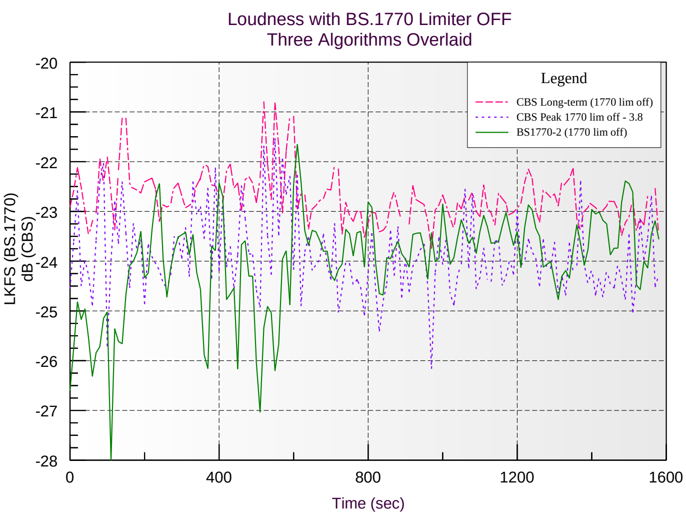

Figure 3-22: Loudness-controlled audio without BS.1770 safety limiter active: output of the CBS and BS.1770 loudness meters in each 10-second interval as a function of time

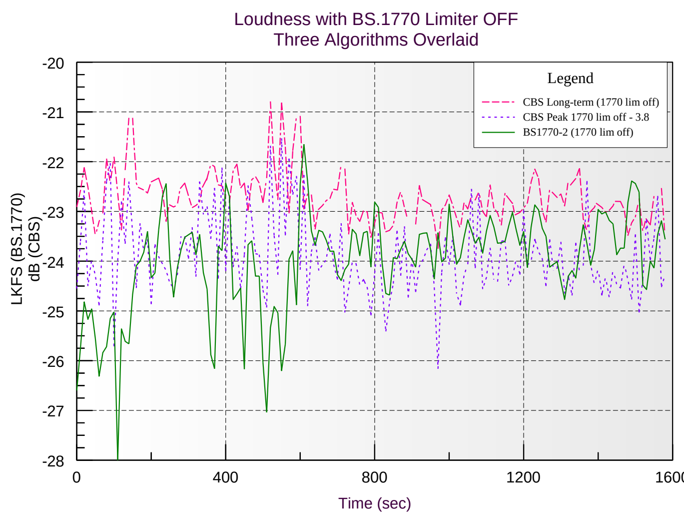

Figure 3-23: Loudness-Controlled Audio With BS.1770 Safety Limiter Active: Output of the CBS and BS.1770 Loudness Meters in each 10-second Interval as a Function of Time

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[10] In the first published version of the paper, we observed the similar dips in the BS.1770-1 (ungated) loudness and hypothesized that they were caused by lack of gating on silence and low-level material. For this reason, we were surprised that BS.1770-2 gating made little difference in the measurements of this material.

[11] Refer to the scatter plots in Figs. 11, 12, and 13 of the ITU-R BS.1770-2 standard.

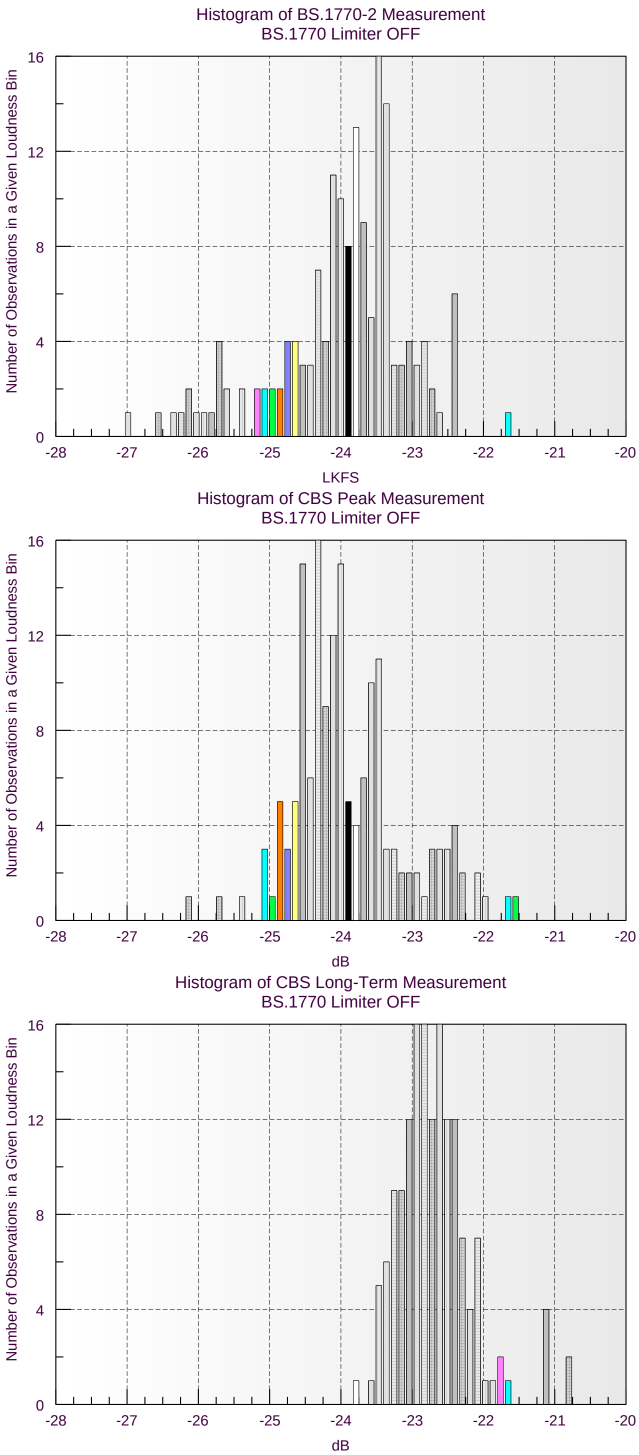

Figure 3-24: Loudness-Controlled Audio With BS.1770 Safety Limiter Defeated: Histograms sorting loudness measurements into 0.25 dB bins

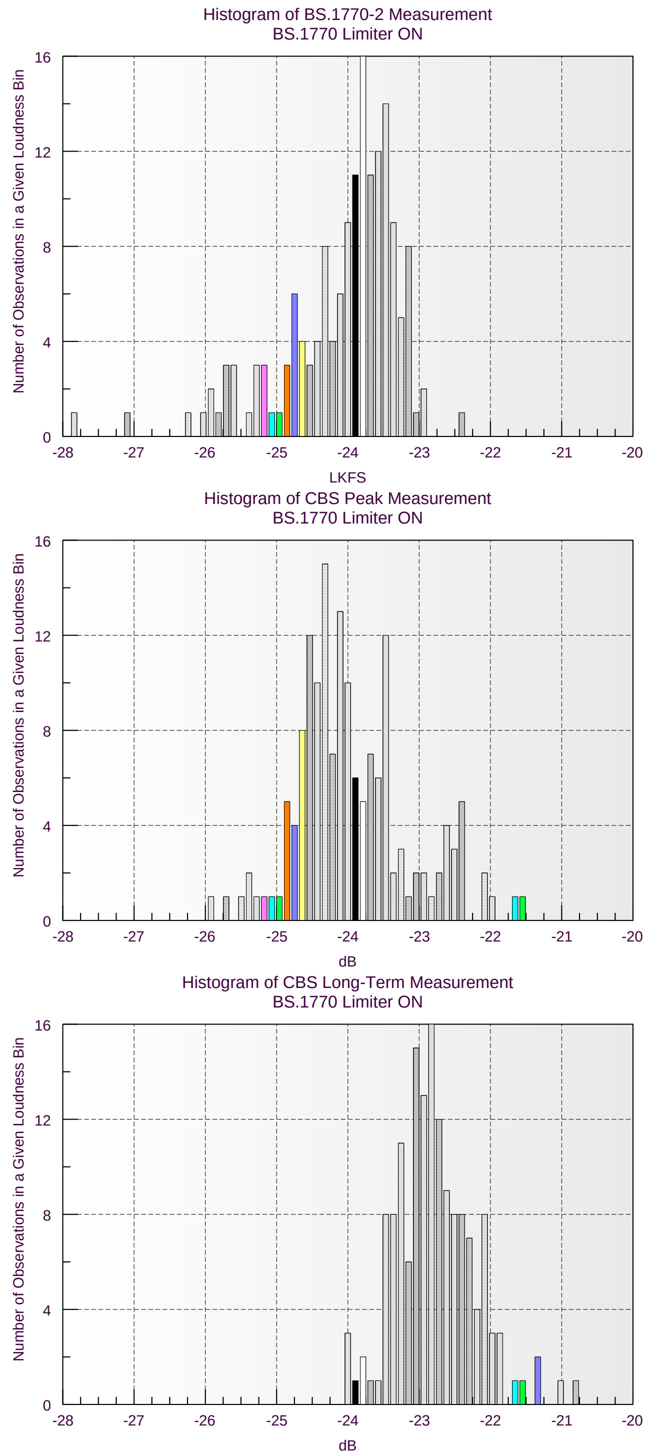

Figure 3-25: Loudness-Controlled Audio With BS.1770 Safety Limiter Active: sorting loudness measurements into 0.25 dB bins

We hypothesize that the fact that the worst-case error of the BS.1770 meter was substantially larger than that of the CBS meter is caused by the BS.1770's meter's not modeling loudness summation or the loudness integration time constants of human hearing.

BS.1770-4 states: "It should be noted that while this algorithm has been shown to be effective for use on audio programmes that are typical of broadcast content, the algorithm is not, in general, suitable for use to estimate the subjective loudness of pure tones." We have noted that the meter tends to over-indicate the loudness of program material that had been subject to large amounts of "artistic" dynamic compression, as is often done for commercials and promotional material — in other words, the meter over-indicates the loudness of program material having an unusually low peak-to-average ratio, which, at the limit, approaches the peak-to-average ratio of a pure tone. We have encountered heated complaints by mixers[12] and producers who stated that such material, when "matched" to the loudness of the surrounding program material via the BS.1770 meter, is considerably quieter in subjective terms. In turn, this has constrained the ability of producers to specify the type of audio processing they had previously used to give this material excitement and punch. We hypothesize that this problem is related to the fact that BS.1770 does not accurately indicate the loudness of pure tones.

Some studies have indicated that when people are asked to assess the loudness of a given piece of material, they state that it sounds louder when underscoring or effects are added to constant-level dialog. The EBU has used these studies to justify the position taken in R128 that a listener's impression of total loudness is more important than dialog level[13]. In our opinion, this misses the point. A more relevant question is whether viewers would want to turn down their volume controls to make dialog quieter when underscoring and effects appear. (In other words, whether effective TV commercial loudness control requires nothing more than applying gain control to commercials such that the BS.1770-2 "short-term" loudness[14] is always limited to 0 Lk.) Regarding this, Orban and Dolby Labs hold similar views. We believe that dialog is the most important element in most television audio and that listeners do not want to turn down their volume controls every time that underscor-

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[12] For example: "I did a –24 [LkFS] piece for Fox that was wall to wall singing and music for two minutes. Because of the overall loudness and continued full audio signal I had to bring it down and when it aired, it was 3 db too quiet even though it matched the magic LkFS number. I have no problem using these meters or meeting specs but they are faulty."
—"wheresmyfroggy," AVID board, 3-28-2011

[13] Dash, Ian; Bassett, Mark; Cabrera, Densil, "Relative Importance of Speech and Non-Speech Components in Program Loudness Assessment," AES Convention Paper 8043, 128th AES Convention (May 2010).

[14] EBU R128 specifies short-term loudness as a BS-1770-1 (ungated) measurement with a three-second integration time.

ing or effects appear under the dialog. The popular Dolby LM100 Loudness Meter[15] in its current revision uses the same Leq(RLB) algorithm as BS.1770 but adds gating to eliminate non-speech material, including silence.

The author has used the Dolby LM100 to measure the output of the Orban processor with a wide variety of speech material, and has observed that this material is almost always controlled within a ±1 dB window as measured on the LM100. In the author's opinion, this demonstrates the benefits of a dialog-centric measurement. Moreover, the author believes it is unwise to rely on a BS.1770 measurement to set the on-air loudness of unadorned dialog because this can cause the dialog to be too loud with respect to other material. The author has experimented with "inverse short-term BS.1770 loudness control" and believes that it sounds unnatural, pumping dialog loudness up and down in a subtly inartistic way as underscoring and effects come and go.[16]

Moreover, the author believes it is unwise to rely on a BS.1770 measurement to set the on-air loudness of unadorned dialog because this can cause the dialog to be too loud with respect to other material. This is consistent with the Audio Engineering Society's "Recommendation for Loudness of Audio Streaming and Network File Playback" (2015)[17], which takes "genre" into account in the context of BS.1770:

> Within a given program, the largest perceived difference to be noted is speech versus music. Speech normalized to the same Integrated Loudness as a music stream inevitably sounds too loud. It is recommended to normalize speech (dialog) segments within other segments 2 to 4 LU (or more) below the loudness of the other segments

The author has experimented with "inverse short-term BS.1770 loudness control" and believes that it sounds unnatural, pumping dialog loudness up and down in a subtly inartistic way as underscoring and effects come and go. Nevertheless, because some organizations require program loudness to be constrained to a given BS.1770 long-term meter reading regardless of how a human listener hears the loudness of

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[15] http://www.dolby.com/professional/products/broadcast/test-and-measurement/lm100.html

[16] See Begnert, Fabian; Ekman, Håkan; Berg, Jan, "Difference between the EBU R-128 Meter Recommendation and Human Subjective Loudness Perception," AES Convention Paper 8489, 131st AES Convention, (October 2011). This paper states, "These loudness-equalized signals gave rise to a perceived maximum loudness difference of 2.8 dB." This is very close to the 3 dB number that has come up in other discussions (such as the one quoted in footnote 12 on page 3-89). While the authors of this paper consider 3 dB to be insignificant, others do not necessarily share this view, particularly advertisers who hear their expensive commercials aired 3 dB quieter than surrounding program material!

[17] Audio Engineering Society: AES TD1004.1.15-10 "Recommendation for Loudness of Audio Streaming and Network File Playback" (2015)

such material, we deemed it wise to provide users with a "BS.1770 safety limiter" in the loudness controller used in the studies in this paper.

The BS.1770 safety limiter is located after the J&T loudness controller in the signal path; we call the combined J&T and BS.1770 algorithms "Orban Loudness Control." It provides "inverse BS.1770" gain reduction for material that exceeds a preset threshold, but it uses an attack integration time of 10 seconds and a release integration time of 3 seconds, so the gain reduction is not a strict inverse of the BS.1770 meter (which uses symmetrical attack and release integration times). The limiter's asymmetrical attack and release integrations times are intended to minimize downward gain pumping on highly compressed material (like commercials) while still allowing the gain to recover quickly when dialog reappears.

The limiter's temporal asymmetry can cause the BS.1770 indication to be slightly higher than the preset target loudness, as can be seen in Figure 3-22 and Figure 3-23. Figure 3-22 and Figure 3-23 show the measured BS.1770 and J&T loudness with and without inverse BS.1770 gain reduction (set to a threshold of –24 LUFS) as a function of time, while Figure 3-24 and Figure 3-25 show the associated histograms, measured with both the BS.1770-2 and CBS meters.

In the BS.1770 measurements in both Figure 3-24 and Figure 3-25, the BS.1770 loudness histogram appear centered on the target loudness (–24 LUFS). However, Figure 3-25 shows that the BS.1770 safety limiter prevented the BS.1770 meter from indicating more than 1 LU higher than the target loudness except for one bin, while Figure 3-24 shows that defeating the safety limiter causes a significant number of BS.1770 loudness events up to be up to 2 LUFS higher than the target loudness. While it would have been trivially easy to make the BS.1770 safety limiter a literal inverse of the meter (thus preventing the BS.1770 loudness from ever indicating above the target loudness) this sounded substantially less natural than using asymmetrical attack and release. Moreover, we still prefer the sound of the J&T loudness controller without any BS.1770 limiting at all. (Note that the CBS readings showing the effect of the BS.1770 Safety Limiter do not show as consistent an affect as the BS.1770 readings. This is because of the differences between the two algorithms.

Studies indicating that BS.1770 is inaccurate at very low frequencies

Another weakness of BS.1770 is that, unlike the CBS loudness controller and meter as implemented in Orban products, the BS.1770 algorithm does not take into account the loudness contributed by the LFE channel, for good reason. Nacross and Lavoie[18] tried to extend the BS.1770 algorithm to include the LFE channel by summing the K-weighted LFE channel's power into the current BS.1770 algorithm, where the gain is weighted for the fact that LFE channel receives a 10 dB gain boost on playback, per Dolby's standards. This modified BS.1770 algorithm failed to agree with the judgments of a subjective listening panel unless a 10 dB attenuation

[18] Norcross, Scott G; Lavoie, Michel C.," Investigations on the Inclusion of the LFE Channel in the ITU-R BS.1770-1 Loudness Algorithm," AES Convention Paper 7829, 127th AES Convention (October 2009)

"fudge factor" was applied to the LFE channel prior to its power summation with the other channels. Nacross and Lavoie concluded:

> A problem exists however, should ITU-R BS.1770 be modified to simply include an attenuated version of the LFE channel. Because the LFE channel receives a 10 dB boost on playback, the low-frequencies on this channel would contribute differently to a loudness measure if they were moved to one of the other main channels, even though the perceived loudness would not appreciably change. This suggests that while LFE content does contribute to the perceived loudness, Equation (2)[19] does not sufficiently predict how that content should be included.

An Australian study may shed light on the failure of BS.1770 when program material contains considerable energy at very low frequencies.[20] The authors used octave-band noise in subjective listening tests with the goal of verifying the K-weighting curve used in BS.1770. The authors state:

> Comparison of the test results with an image of the filter curve currently specified in ITU-R Recommendation BS.1770 (Figure 13) shows good agreement at 250 Hz and above 500 Hz, reasonable agreement at 500 Hz, but marked difference in the bottom two octaves.
>
> The relatively good performance of the BS.1770 algorithm in ITU trials suggests that, in partial loudness terms, there was probably not much test content in the 125 Hz band or below. While the existing BS.1770 filter curve is probably a good choice in applications where the program is dominated by speech, and it is certainly an improvement on the A and B curves in that application, it is likely to give significant errors in measuring the loudness of other programs with more partial loudness in the lower frequencies, such as movie soundtracks and popular music. It is therefore desirable to improve on this filter for more general measurement of program loudness.

Discussion and Conclusions

Several studies have shown that the loudness "comfort range" for typical television listening is +2, −5 dB[21]. Beyond this range, a viewer is likely to become annoyed, eventually reaching for the remote control to change volume (or worse from the broadcaster's point of view, to mute a commercial). Whether measured via the CBS or BS.1770 algorithms, the CBS loudness controller algorithm in Orban's current

---

19 $$Leq(w) = \left[\frac{1}{T} G_{LFE} \int_T^0 \frac{x_w^2}{x_{ref}^2} dt + \sum_i \frac{1}{T} \int_0^T \frac{x_{w,i}^2}{x_{ref}^2} dt \right], dB$$

$$i = L, R, C, L_s, R_s$$

20 Cabrera, Densil; Dash, Ian; Miranda, Luis, "Multichannel Loudness Listening Test," AES Convention Paper 7451, 124th AES Convention (May 2008)

21 ATSC A/85:2009 Annex E, "Loudness Ranges"

products effectively controls subjective loudness to much better than this +2, −5 dB window.

In the original version of this paper, we had assumed that results using BS.1770 metering would be more consistent if that algorithm employed gating to prevent unadorned dialog from reading low compared to music and dialog with substantial background music or effects. However, this did not prove to be true with the program material we used for testing—the results from the BS.1770-1 (ungated) and BS.1770-2 (gated) measurements were similar when measuring material that had been processed by the CBS Loudness Controller. It is likely that the loudness-controlled material seldom caused the gate to act. (The CBS algorithm does not need silence gating because it is a "short-term" loudness measurement that incorporates cascaded models of the "instantaneous" and "short-term" loudness time constants of human hearing[22], which the BS.1770 algorithm does not.)

Controlling loudness to a standard such as BS.1770 says nothing about the subjective acceptability of the loudness controller's action. We have found that a simple loudness controller that uses the inverse of the BS.1770 short-term meter's output to control loudness by gain reduction can cause unnatural-sounding gain pumping of dialog when underscoring and effects appear under the dialog. More complex automatic loudness controllers can produce all of the well-known artifacts of dynamics processing, including noise breathing, spectral inconsistency, gain pumping, and harshness. Improperly designed multiband compressors can reduce dialog intelligibility[23]. This is why it is important to carefully assess the audio quality and side effects that an automatic loudness controller produces so that one can choose a device that controls loudness effectively without producing objectionable and unnatural artifacts that can fatigue audiences. Different loudness controllers do not provide equally good subjective results even if they produce identical measurements on a loudness meter.

Based on extensive experimentation with typical broadcast material, we believe that the CBS loudness meter locks onto dialog more effectively than does BS.1770, particularly when the dialog is accompanied by underscoring and/or effects. Accordingly, the CBS Loudness Controller in Orban products, which uses the CBS loudness metering algorithm as its core loudness reference, produces consistent and naturally balanced dialog levels regardless of the program material and mixing style. Unlike the BS.1770 meter, the CBS technology does not unnaturally penalize material having a low peak-to-RMS ratio, so it allows mixers and producers to freely use "artistic

[22] For example, see Glasberg, B.R. & Moore, B.C.J. (2002) "A Model of Loudness Applicable to Time-Varying Sounds," J.AES, vol.50:5, pp.331-342, May 2002.

[23] Stone, Michael A.; Moore, Brian C. J.; Füllgrabe, Christian; Hinton, Andrew C., "Multichannel Fast-Acting Dynamic Range Compression Hinders Performance by Young, Normal-Hearing Listeners in a Two-Talker Separation Task," J. AES Volume 57 Issue 7/8 pp. 532-546; July 2009

compression"[24] and other well-established production techniques with the knowledge that such material will be neither too loud nor too quiet when compared to the surrounding program.

In the Orban Loudness Controller, the BS.1770 Safety Limiter (which follows the J&T controller) can be used to any extent desired by setting its threshold higher than the target loudness. We believe it is mainly useful for organizations that rely on logging the BS.1770-4 loudness to verify compliance with international regulations, and that setting its threshold 2 LU above the target loudness minimizes inconsistent subjective loudness caused by "genre" issues as discussed in AES TD1004.1.15-10.

—Robert Orban, revised February 2020

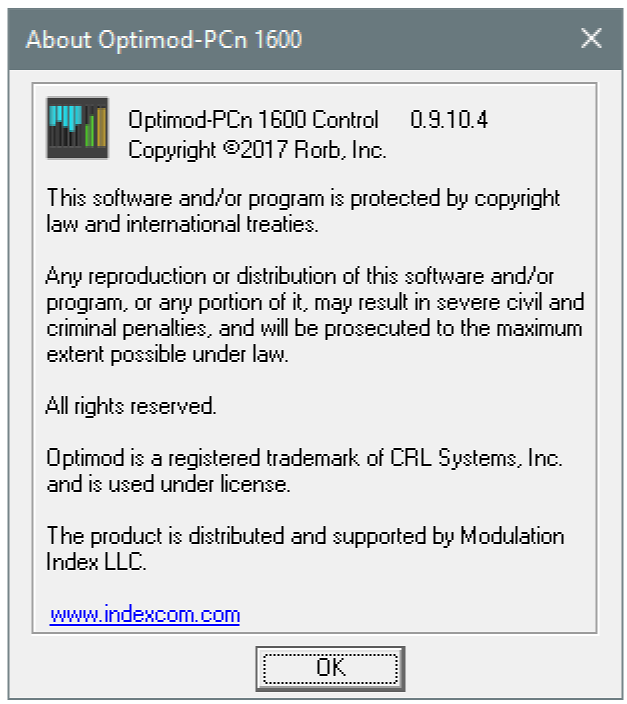
[24] It appears that the group that created R128 may be biased against this style of production: "Again, this does NOT mean that within a programme the loudness level has to be constant, on the contrary! It also does NOT mean that individual components of a programme (for example, pre-mixes or stem-mixes, a Music & Effects version or an isolated voice-over track) have all to be at the same loudness level! Loudness variation is an artistic tool, and the concept of loudness normalisation according to R128 actually encourages more dynamic mixing!" EBU TECH 3343, op. cit., p. 17

Section 4

Software Summary

OPTIMOD-PCn Software Summary

The OPTIMOD-PCn software version number is available through the Control Application `Help > About` menu.



Application Installation File

`Setup1600xxx.exe` – Installer:

- Installs the Control Application– `Optimod3.exe`
- Installs the Agent – `OptimodAgent3.exe`
- Installs the Service – `OptimodService3.exe`
- Installs the factory presets
- Installs run-time libraries
- Detects whether the computer's CPU supports the Intel AVX2 vector instruction set and installs support for AVX2 if appropriate.

Service

- Implements the audio signal processing and provides hooks into the Windows sound I/O system.
- Provides remote control security and access.
- Starts 120 seconds after Windows finishes booting. This delay allows Windows to settle down so that it does not compete with the Service for CPU cycles. When it starts automatically, the Service will load the same processing presets that were active on the OPTIMOD-PCn Processors when the computer shut down, and will start processing audio.
- Supports the API
- Supports SNMP.

SNMP is not supported in v0.9 software.

- **Authorization**: If your version of 1600PCn requires a hardware key, the Service will only start up if it can determine that the USB security key is plugged into an available USB socket. The key is driverless; no driver installation is necessary.

  If your 1600PCn requires CodeMeter software authorization, this must be correctly installed and authorized. See the document *Instructions to create license request file for Optimod PCn.pdf*, which the 1600PCn installer copied to the service computer.

Agent (Tray Icon)

- The OPTIMOD-PCn Agent is a tray icon that allows you to launch the OPTIMOD-PCn Control Application from the Windows System Tray.
- If you right click the tray icon, a menu pops up with a checkable item named AUTO-START THE TRAY ICON. (The "Auto-start the tray icon" item will be checked by default.)
- If you uncheck AUTO-START THE TRAY ICON, the tray icon will not re-appear when you re-start the PC.
- To re-engage the icon, go to `Start/Programs/Orban/OPTIMOD-PCn 1600` and select `OptimodPcAgent`.

You do not need to open the OPTIMOD-PCn Control Application to toggle the auto-start feature on and off.

1600PC Control Application

- Provides access to all OPTIMOD-PCn I/O Mixer controls, DSP processing parameters, and presets.
- Can be used as a client to remotely access OPTIMOD-PCn Processors running on computers other than those doing the audio processing.
- Is not copy-protected and can be installed on as many computers as you wish.
- It is not necessary (or desirable) for the Service to run on a computer that does not do audio processing but only hosts 1600PC. (The OPTIMOD-PCn installer will not install the Service if you tell it during the install process to install only the remote client.)

Registry

- I/O Settings
- Active Preset
- Encrypted Security
- Application GUI Settings

Section 5

Control API

OPTIMOD 1600PCn Control API

The OPTIMOD 1600PCn Service application hosts a TCP/IP terminal server to allow external control of OPTIMOD 1600PCn processors from either a Telnet client or a custom third party application. All commands are simple text strings. Upon receiving valid commands, OPTIMOD 1600PCn will confirm by returning a simple text string status message. By implementing external control this way, multiple OPTIMOD 1600PCn Processors can be controlled using standards-based network protocols (that are not Microsoft Windows-specific) anywhere that network connectivity is available.

To protect the Terminal Server from unauthorized remote access, the default setting for the control "Use localhost only for Terminal Access" is enabled. In this case, only Telnet connections from the localhost address 127.0.0.1 on the default port 12101 are accepted. Disabling this will allow Telnet connections via Ethernet from other computers, using the IP address of the host computer. This control is located in the OPTIMOD 1600PCn remote application in the TOOLS>SERVICE SETUP dialog box and is accessible only when the application is disconnected from a 1600PCn processor (i.e. when controls and meters are not displayed). The default port can be changed here too.

To control OPTIMOD 1600PCn externally, establish a Telnet connection and issue commands and parameters, either by typing them directly into a Telnet client or by placing them within batch files. Then process them with a scriptable Telnet client that supports this operation, such as PuTTY, along with its companion command-line interpreter, Plink. Both of these applications are available for free download. Search "PuTTY" with a search engine like Google to find a download site.

Automating control changes is possible by using the Windows Task Scheduler to launch batch files at the desired time.

Custom third party applications can be developed to use this protocol. Additionally, you can include this protocol in an existing application by using small subsets of the standards-based Telnet protocols directly, or for simplicity, by using scripting or by calling batch files with a Telnet client such as PuTTY along with its companion command-line interpreter, Plink. Scripting eliminates the need to develop networking code or otherwise contend with complex, limited-function Microsoft Windows specific programming APIs. Developing a third party application or including the protocol in an existing application eliminates having to install and configure additional applications. Using small subsets of the standards-based Telnet protocol allows more operational flexibility without a system performance hit and simplifies development compared to proprietary OEM APIs.

Table 5-1 on page 5-2 describes the available top-level command strings and their functions. To demonstrate this functionality, you may type the commands directly into a Telnet client terminal and see OPTIMOD 1600PCn confirmation and status messages. The lists of addressable controls are found in *Table 5-2: Processing Controls Available from API* on page 5-8 and *Table 5-3: Settings Controls Available from the API* on page 5-9.

Commands are case-sensitive and must be entered exactly as shown, including spaces. Replace "preset name" and "setting name" with the actual Preset name and Settings name respectively. Replace `Processor x` with the name of the processor you are addressing; for example, `<Processor 1>`. Note that if you have created an alias name for a Processor, you cannot use this in the API; use the actual "Processor" name instead.

The file processing commands `fpinput` and `fpoutput` are explained in *Using the API for File Processing* starting on page 3-84.

Administering through Ethernet TCP/IP

Using a terminal program like PuTTY, you can control an OPTIMOD 1600PCn Processor through a network Ethernet TCP/IP connection to the computer that hosts the 1600PCn you wish to access. An ASCII programmable GPIO device, such as a Broadcast Tools SRC-16, with an appropriate Serial-Ethernet server, can also be used for

Command	Function
`help`	Lists important API commands
`version`	Retrieves control application version information.
`plist`	Returns a list of all processing controls available to the API.
`slist`	Returns list of all controls available to the API in Settings (configuration).
`header`	Returns all control information formatted to be included in a C program file.
`fpinput [path]`	Specifies a file or folder to be processed using file processing functionality
`fpoutput [path]`	Specifies the destination folder for file processing
`disconnect`	Disconnects Telnet connection.
`<Processor x> pstatus`	Returns values for each control in the active preset.
`<Processor x> sstatus`	Returns values for each control in the active Settings configuration.
`<Processor x> AP ??`	Retrieves all available Presets.
`<Processor x> AP ?`	Retrieves Active Preset.
`<Processor x> RP preset name`	Recalls Processing Preset.
`<Processor x> DP preset name`	Deletes a preset.
`<Processor x> AS ??`	Retrieves all available Settings files.
`<Processor x> AS ?`	Retrieves Active Settings file.
`<Processor x> RS settings name`	Recalls a Settings file.

Table 5-1: Top-Level Telnet Command List

hardware control and status. Multiple connections and devices can be used. When you are connected, you can retrieve information and change settings on the 1600PCn by using simple ASCII commands as discussed in *OPTIMOD 1600PCn Control API* starting on page 5-1.

- Valid commands are in either upper or lower case, not a combination.
- Only one valid command is permitted per line.
- OPTIMOD 1600PCn will not respond to unrecognized commands.
- The character code supported is ASCII.

Connecting via TCP/IP Using PuTTY on a PC

- You can use the 1600PCn with any computer or terminal that is compatible with TCP/IP networking.
- The OPTIMOD 1600PCn Service Settings determine network connection security:

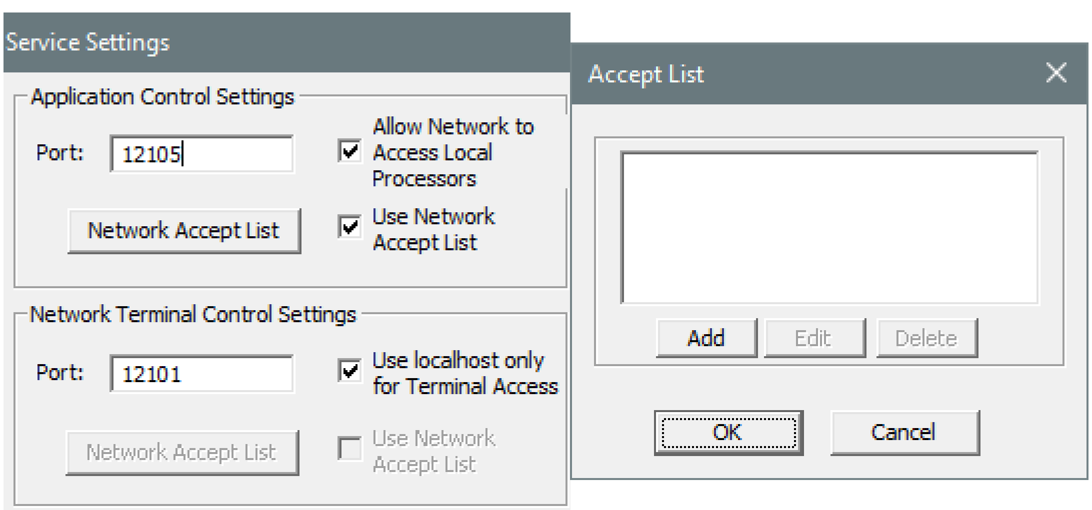
  A) Using the OPTIMOD 1600PCn Control Application, select TOOLS > SERVICE SETTINGS.

  B) Assign an available TCP/IP Port. (Default is 12101.)

  C) If you will be communicating from a computer other than the computer hosting the Service, uncheck "Use localhost only for Terminal Access."

  D) (Optional) Check "Use Network Accept List" and add allowed IP addresses to list. (Step 8 on page 2-7 has more detail.)

Because the API does not use passwords in the individual commands, it is important to use the "Use Network Accept List" feature to enforce security if you wish to communicate from a computer other than the computer hosting the Service.



- Connect your computer or terminal with an Ethernet cable:

A) Connect an appropriate Ethernet cable on your terminal, computer, or device to either an Ethernet port on the XPM-AM's host computer or the Ethernet network that will allow access the 1600PCn host computer.

- If connecting to a standard Ethernet network, use a standard Ethernet cable and TCP/IP network connection.
- If connecting directly to another computer, use a crossover Ethernet cable.

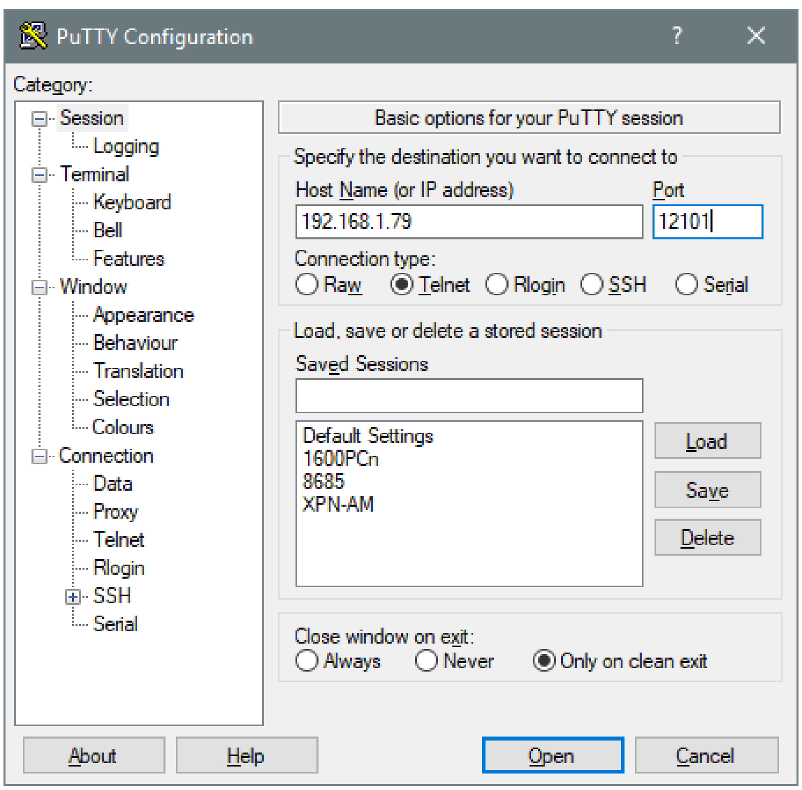
When a crossover Ethernet cable is used, at least one computer must have a static IP address. The other computer may have a static address or may have DHCP enabled.

To enable DHCP in a given computer, its Internet Protocol (TCP/IP) Properties must be set to "Obtain an IP address automatically." Internet Protocol (TCP/IP) Properties is also where you set a computer's static IP address.

In either case, when using a crossover cable, both computers should be assigned the same IP subnet unless special routing has been configured.

B) Start PuTTY

The CONFIGURATION dialog box appears.



C) Set the Host Address to that of the OPTIMOD 1600PCn Service host computer.

D) Set the Port Number to that of the OPTIMOD 1600PCn Service host computer, (Default is 12101).

E) Select TELNET as the CONNECTION TYPE.

F) (Optional) Type a name for your connection in the SAVED SESSIONS box and click SAVE.

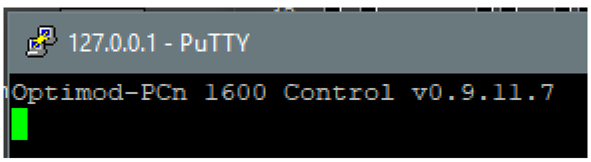
G) Click OPEN.

The 1600PCn will immediately connect and the 1600PCn connect banner will display:

```
Orban OPTIMOD 1600PCn Control vx.x.x.xx
```

Press your computer's ENTER key once. Commands may now be sent and status received.

OPTIMOD 1600PCn Command and Status Reference is in *OPTIMOD 1600PCn Control API* starting on page 5-1.



Using the API: Examples

These examples show how PuTTY and Plink can be used to control OPTIMOD 1600PCn using scripting files on a local or remote computer as the OPTIMOD 1600PCn to be controlled. Plink and all associated scripting text, PuTTY, and .cmd files should be located together in the same user-defined directory unless the path is specified in the .cmd files.

In the example, replace "127.0.0.1" with the IP address of the computer running the Service that implements the controlled Processor and "12101" with the card's port number. 12101 is the default; see step 8 on page 2-7.

127.0.0.1 is "localhost" and is used if you are running the Service on the same computer you are using to connect to the API.

Each control session requires two ASCII files and one optional Shortcut file:

- a .cmd file that calls Plink to establish a Telnet connection to OPTIMOD 1600PCn
- a reference.txt file that contains the actual control script.
- an optional Windows Shortcut .lnk file calls the .cmd file with a suppressed Command Box.

Recalling the CHR MX MED preset

These two files will recall the CHR MX MED preset.

The file "recall_CHR MX MED_preset.cmd" contains:

```
plink -telnet -P 12101 127.0.0.1 < recall_CHR MX MED_preset.txt
```

The file "recall_CHR MX MED_preset.txt" contains:

```
<Processor 1> RP CHR MX MED
disconnect
```

Automated Batch-Processing of all Files in a Specified Directory

The two files described below will process all files in the d:\temp\music_source directory, placing them in the d:\temp\music_destination directory.

> Because in this case the command file is run on the same computer that hosts the Optimod Service, the file uses a "localhost" (127.0.0.1) connection to Port 12101. If you want to run the cmd file on another computer on your LAN, refer to step 8 on page 2-7. To enforce security, it is wise to exploit the Optimod's Network Terminal Control Settings "Network Accept List" feature to limit access to computers having a specified IP address [see step (D) on page 5-3].

The file "batch_proc.cmd" contains:

```
"C:\Program Files\PuTTY\plink" -batch telnet://127.0.0.1:12101<batch_proc.txt
EXIT /b
```

The file "batch_proc.txt" contains the following. (Note that the empty lines terminated with CR/LF are needed to handle the messages that the Optimod API returns.)

```
[CR/LF character]
fpinput d:\Temp\music_source
[CF/LF character]
fpoutput d:\Temp\music_destination
disconnect
```

When you run batch_proc.cmd, the Windows Command Prompt box displays:

```
d:\Temp\Tone_Source>"C:\Program Files\PuTTY\plink" -batch [line wrap added]
0< telnet://127.0.0.1:12101 0<plink_test.txt
Optimod-PCn 1600 Control v0.9.11.7
FP_INPUT:source folder found
FP_OUTPUT:destination folder found. File Processing started.
```

Because of the presence of `EXIT /b` in the .cmd file, the command file will terminate before it can echo the Optimod's message indicating that all files were processed:

`FP_STATUS:folder processed successfully`

If you want to see this message in the command box, omit `EXIT /b` from the .cmd file. However, you will then have to terminate the command processing manually via CTRL C from your keyboard.

Processing Controls Available From the API

A vast majority of users will have no reason to adjust processing controls from the API because it is much more convenient to do it from the 1600PCn PC application. The processing controls API is mainly useful for developers who wish to develop a custom interface.

The valid values for a given control correspond to the available values of the control as displayed in the 1600PCn's GUI (PC application). You must use the identical format, including whether a decimal point is included, and if it is, how many decimal places are displayed. For example, 12, 12.0, and 12.00 are not equivalent because the values are parsed as strings and not as numerical values. You can tell if you have entered a valid value because the API will return an acknowledgement. To fetch a list of all valid values, type the control name followed a space and two question marks (see the example below).

The HEADER command returns a list of all controls (both Processing and Settings) and their valid values. To capture the header to a text file, refer to the PuTTY command CREATING A LOG FILE OF YOUR SESSION in PuTTY help.

Example to change the control value:

Type the following, including spaces, exactly as shown below:

```
<Processor 1> ST_AGC On
```

(returns)

```
ST_AGC:On
```

Example to query the control value:

Type the following:

```
<Processor 1> ST_AGC ?
```

(returns)

```
ST_AGC:On
```

Example to fetch the valid settings of a control:

Type the following:

```
ST_AGC ??
```

(returns)

```
ST_AGC
On
Off
```

As per the example above, all commands in the tables below must be preceded by `<Processor x>[space]`, where "x" is the name of the Processor you are addressing, and [space] is one space without brackets. Valid Processor numbers are 1 through 16, depending on the number of processors specified in the NUMBER OF PROCESSORS field in TOOLS>SERVICE SETTINGS, which is limited by the number of available Processors authorized by your security key or CodeMeter authorization. You can view the number of available Processors in the PROCESSOR CONFIGURATIONS AVAILABLE field in TOOLS>SERVICE SETTINGS.

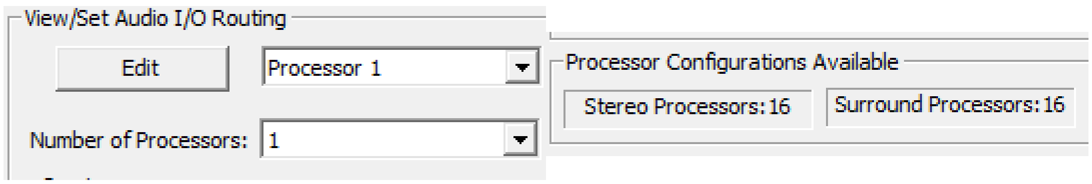


Table 5-2: Processing Controls Available from API

Processing Controls Available from the API	(alphabetical order)
ALLF_PEQ_IN_OUT	ST_B4_BREAKPOINT
ALL_EQ_IN_OUT	ST_B4_COMP_KNEE
B1B2_XOVER_SLOPE	ST_B4_COMP_RATIO
BRILLIANCE_IN_OUT	ST_B4_COMP_THRSH
COMPRESSOR_MODE	ST_B4_DELTA_REL
COMP_THRESH_OFFSET	ST_B4_DIFF_GR
CROSSOVER_MODE	ST_B4_LIMIT_ATTACK
HF_ENHANCER_IN_OUT	ST_B4_ON_OFF
HF_PEQ_IN_OUT	ST_B4_OUTPUT_MIX
LF_PEQ_IN_OUT	ST_B5_ATTACK
LF_SHELF_IN_OUT	ST_B5_BREAKPOINT
MF_PEQ_IN_OUT	ST_B5_COMP_KNEE
MONO_BASS	ST_B5_COMP_RATIO
MONO_BASS_XOVER	ST_B5_COMP_THRSH
MX_BASS_LIMIT	ST_B5_DELTA_REL
MX_BASS_PRELIM	ST_B5_DIFF_GR
MX_BASS_PRELIM_MODE	ST_B5_DWNWRD_EXP
MX_CLIP_DIST	ST_B5_LIMIT_ATTACK
MX_CLIP_THR	ST_B5_ON_OFF
MX_OVERSHOOT_LIMIT_MODE	ST_B5_OUTPUT_MIX
MX_SP_BASS_PRELIM	ST_BAND_21_COUPL
MX_SWITCH	ST_BAND_23_COUPL
PEQ_ALL_FREQ	ST_BAND_32_COUPL
PEQ_ALL_GAIN	ST_BAND_34_COUPL
PEQ_ALL_WIDTH	ST_BAND_45_COUPL
PHASE_CORRECTOR	ST_BASS_CLIP
PHASE_CORRECT_XOVER	ST_BASS_CLIP
SENSITIVITY_TRIM	ST_BRILLIANCE
SE_AMOUNT	ST_BS1770_LDNES_CTRL_THR
SE_IN_OUT	ST_DE_STEREO_COUPL
SE_RATIO_WIDTH	ST_DJ_BASS_BOOST
STEREO_SYNTH_ALG	ST_DWNWRD_EXP
STEREO_SYNTH_MODE	ST_FINAL_LIMIT_DRIVE
STEREO_SYNTH_SEPARATION	ST_HF_ENHANCER
ST_ACG_RELEASE	ST_HIGH_PASS
ST_AGC	ST_HIGH_PASS_Q
ST_AGC_BASS_ATTACK	ST_LDNES_CTRL_ATTACK
ST_AGC_BASS_COUPLING	ST_LDNES_CTRL_BASS_CPL
ST_AGC_BASS_RELEASE	ST_LDNES_CTRL_THR
ST_AGC_BASS_TH	ST_LESS_MORE
ST_AGC_DIFF_GR	ST_LOW_BASS_FREQ

Processing Controls Available from the API	(alphabetical order)
ST_AGC_DRIVE	ST_LOW_BASS_GAIN
ST_AGC_GATE_THRESH	ST_LOW_BASS_Q
ST_AGC_IDLE_GR	ST_LOW_PASS
ST_AGC_MASTER_ATTACK	ST_LOW_PASS_EQ
ST_AGC_RATIO	ST_LOW_PASS_EQ_Q
ST_AGC_WINDOW_RELEASE	ST_MB_DRIVE
ST_AGC_WINDOW_RELEASE	ST_MB_GATE_THR
ST_AGC_WINDOW_THRESH	ST_MB_RELEASE
ST_AGC_WINDOW_THRESH	ST_MUSIC_SPEECH
ST_B12_CROSSOVER	ST_PASSTHRU_GAIN
ST_B1_ATTACK	ST_PASSTHRU_SW
ST_B1_BREAKPOINT	ST_PEQ_HIGH_FREQ
ST_B1_COMP_KNEE	ST_PEQ_HIGH_GAIN
ST_B1_COMP_RATIO	ST_PEQ_HIGH_WIDTH
ST_B1_COMP_THRSH	ST_PEQ_LOW_FREQ
ST_B1_DELTA_REL	ST_PEQ_LOW_GAIN
ST_B1_DIFF_GR	ST_PEQ_LOW_WIDTH
ST_B1_LIMIT_ATTACK	ST_PEQ_MID_FREQ
ST_B1_ON_OFF	ST_PEQ_MID_GAIN
ST_B1_OUTPUT_MIX	ST_PEQ_MID_WIDTH
ST_B2_ATTACK	ST_PHASE_ROTATOR
ST_B2_BREAKPOINT	ST_SPEECH_HPF
ST_B2_COMP_KNEE	ST_SPEECH_HPF
ST_B2_COMP_RATIO	ST_SP_B1_ATTACK
ST_B2_COMP_THRSH	ST_SP_B1_COMP_THRSH
ST_B2_DELTA_REL	ST_SP_B2_ATTACK
ST_B2_DIFF_GR	ST_SP_B2_COMP_THRSH
ST_B2_LIMIT_ATTACK	ST_SP_B3_ATTACK
ST_B2_ON_OFF	ST_SP_B3_COMP_THRSH
ST_B2_OUTPUT_MIX	ST_SP_B4_ATTACK
ST_B3_ATTACK	ST_SP_B4_COMP_THRSH
ST_B3_BREAKPOINT	ST_SP_B5_ATTACK
ST_B3_COMP_KNEE	ST_SP_B5_COMP_THRSH
ST_B3_COMP_RATIO	ST_SP_DWNWRD_EXP
ST_B3_COMP_THRSH	ST_SP_HIGH_PASS_Q
ST_B3_DELTA_REL	ST_SP_MB_RELEASE
ST_B3_DIFF_GR	ST_TRANSIENT_ENHANCE
ST_B3_LIMIT_ATTACK	ST_TRANSIENT_ENHANCE
ST_B3_ON_OFF	SUBHARMONIC_INJECTION
ST_B3_OUTPUT_MIX	SUBHARM_IN_OUT
ST_B4_ATTACK	THRESHOLD_TRIM

Settings Controls Available From the API

The Settings API can be used to automate proof-of-performance tests.

Table 5-3: Settings Controls Available from the API

Settings Controls Available from the API	(alphabetical order)
ALGORITHM	OUTPUT_PRE_E
BS1770_LC_FLAG	PHASE_CORRECT_DEFEAT
BYPASS_GAIN	PROC_PRE_E
EXTERNAL_AGC	RIGHT_TONE
INPUT_REF_LEVEL	SQUARE_FREQUENCY
LEFT_TONE	STEREO_SYNTH_DEFEAT
MAX_LOW_PASS	TARGET_LOUDNESS
METER_DELAY	TEST_FREQUENCY
ORBAN_LC_FLAG	TEST_MOD_LEVEL

OUTPUT_DITHER	TEST_WAVEFORM
OUTPUT_LEVEL	

Section 6

Specifications

Specifications

SYSTEM

Audio Processors: Each Processor in OPTIMOD-PCn realizes a mono, stereo, or surround audio processor. Except for mono, each processor consists of the following cascaded processing elements: Internal Processing: Input → Left/Right Phase Skew Corrector → Mono Bass → Stereo Synthesizer → Stereo Enhancer → Two-Band defeatable AGC with window gating → Equalizer/HF Enhancer → Multiband Compressor → Optimix® upmixer → Peak Limiter → Automatic Loudness Controller → Output. In surround processors, Optimix is located just before the AGC, and all dynamics processing operates on the upmixed signal when Optimix is active. See *Figure 3-2: OPTIMOD-PCn Digital Signal Processing Simplified Block* Diagram on page 3-13.

Number of Audio Processors: Software is priced according to the total number of Processors permitted to run simultaneously on a given computer. The number of Processors realizable on a given computer depends on the processing features used and the power of the computer's CPU. Plan your installation by using the information in *CPU* starting on page 1-4.

Frequency Response: ±0.1dB, 2-20,000Hz (Bypass software running).

Input/Output Delay: Typically 450 to 1000 ms. Varies according to the number of processing features that are activated.

Internal Filters: 10, 11, 12, 13, 14, 15, and 20kHz can be used to provide additional antialiasing for low sample rate services.

Internal Sample Rate: 48-256 kHz, as appropriate.

Input/Output Sample Rate: 44.1, 48, 96, or 192 kHz. 44.1, 96, and 192 kHz use high-quality synchronous sample rate converters built into OPTIMOD-PCn and do not rely on Windows' built-in SRC.

Internal Resolution: 32-bit or 64-bit floating point, as appropriate for the processing being performed.

Input/Output Resolution: 16-bit or 24-bit fixed point. If the output is configured for 16 bits, First-order noise-shaped dither can be applied prior to truncation to 16 or 24 bits from 32-bit float.

Peak Control: Peak limiter is oversampled at 192 (non-MX mode) or 256 kHz (MX mode), yielding a worst-case overshoot of 0.5 dB at the analog output and for all output sample rates. (To achieve this performance at 32 kHz output sample rate, it is necessary to set OPTIMOD-PCn's lowpass filter cutoff frequency to 15 kHz.)

Phase Response: Linear-phase (constant group delay) or allpass, user selectable.

AGC (Automatic Gain Control): ±12dB/24dB gain range, Two-Band, Gate and Window enabled.

Stereo Enhancer: Orban-developed L−R dynamic expansion triggered by L+R transients.

Equalizers: Shelving Low Bass EQ, selectable 6 dB, 12dB or 18dB/octave. Four-band Parametric EQ with analog-style bell-shaped curves. Program Adaptive HF Enhancer. Brilliance control uses Band 5 of the multiband compressor as a steep slope shelving equalizer (12 or 18 dB/octave for linear-phase or allpass crossovers respectively). Sweepable lowpass and highpass filters with adjustable slopes, 6 to 24 dB/octave.

Multiband Compressors: 25dB gain reduction range, Five-Band. Inline and parallel modes available.

Optimix® Upmixer: Upmixes stereo (2.0) to 5.0-channel (Lf/C/Lf/Lsur/Rsur) surround. (See *Optimix® Automatic Stereo→Surround Upmixer* on page 3-15.)

LFE Channel: There is no upmixed LFE channel because Optimix places energy below 80 Hz in the Lf and Rf channels instead of in an LFE channel. This makes it maximally compatible with home theater receivers' bass management functionality—if Lf and Rf speakers are "small," the receiver will route bass to the LFE channel if present.

Phase Corrector: A defeatable multidimensional phase correction for the center channel: can correct multiple, unequal delay skew errors among different program elements common to the L and R input channels.

Upmix Modes: Provides two upmix modes: Wrap (puts the listener "in the middle of the band") and Frontal (puts the listener "in the audience.")

Downmix Compatibility: In Wrap mode and with the phase corrector defeated, the stereo LoRo mixdown of the upmix is identical to the original stereo input regardless of the settings of other user controls (Front/Rear Balance and Center Width). Activating the phase corrector improves the mono downmix compared to the mono downmix of the original stereo. Frontal mode slightly increases ambience in the mixdown.

Auto-detection: In surround instances of OPTIMOD-PCn, Optimix can detect the presence of stereo input material and automatically activate Optimix via a smooth, delay-matched crossfade.

Latency: 212 ms

Limiter: >12dB gain reduction range. Two modes available: low-IM look-ahead (to minimize CPU usage) and MX (to achieve highest performance). Moreover, the MX limiter's overshoot compensator can be operated in Soft mode (for lowest distortion) or Hard mode (to achieve highest loudness without audible gain pumping).

Loudness Controller: Constrains subjective loudness to a user-adjustable threshold via the 1981 Jones & Torick CBS Technology Center algorithm, as further refined and developed by Orban. The algorithm also drives a subjective loudness meter, which is displayed on the 1600's GUI. In dual-mono mode, there are two independent loudness controllers and meters.

BS.1770 Safety Limiter: Constrains Integrated BS.1770-2 (and higher) loudness to preset value, with an attack time of 10 seconds and a release time of 3 seconds.

Pass-through mode: Delay-matched to active processing. Gain adjustable from −10 dB to +10 dB.

Number of Factory Presets: More than 50, each with 19-step LESS-MORE control. Presets are fully customizable.

Number of User Presets: Essentially unlimited. User presets can be saved on the host computer's hard drive or on other storage devices.

Audio I/O

OPTIMOD-PCn uses Windows built-in WASAPI sound kernel for audio input and output, and is compatible with sound devices whose drivers support WASAPI. Compatible with Windows 7 and higher.

COMPUTER

Minimum System Requirements: Recommend Intel CPU and Chipsets will change depending on the current offerings from Intel. We recommend Intel i-series processors with clock speeds of 3 GHz or higher.

Software:

1600PC Control application: provides subjective adjustment controls of the audio processing and remote administration. It also allows factory and user presets to be recalled from and save to a host storage device such as a hard disk drive. The control application client addresses multiple OPTIMOD-PCn Processors.

API:

IP API: provides complete remote administration over TCP/IP. The OPTIMOD-PCn Service application hosts a TCP/IP Terminal Server to allow external control of the OPTIMOD-PCn cards from either a Telnet/SSH client or a custom third party application. All OPTIMOD-PCn Presets and Mixer Controls are accessible and all commands are simple text strings.

Telnet/SSH: RFC 318 compliant basic subset. Compatible with Windows Telnet and PuTTY Telnet clients.

TCP/IP Port: user assignable.

Serial API: provides complete remote administration over Serial port communication. The OPTIMOD-PCn Service application hosts a Serial terminal server to allow external control of the OPTIMOD-PCn cards from either a terminal client or a custom third party application. All OPTIMOD-PCn Presets and Mixer Controls are accessible and all commands are simple text strings.

Terminal Programs: Compatible with terminal programs such as Windows HyperTerminal, and PuTTY VT-100 clients.

COM Port: user assignable from COM1 to COM100.

Status Monitoring: SNMP RFC 1157 compliant. Monitors all Audio Input and Output presence, Mixer Control status, Active Preset, Control Application Connections, and Control Client Connections.

1601PCn and 1602PCn Features

Models 1601PCn and 1602PCn are lower cost versions of 1600PCn software that remove certain features. For a list of removed features, see *1602PCn Features* on page 1-12 and *1601PCn Features* on page 1-12.

Section 7

Index

D

E

F

G

H

I

O

P

T