TDM

AUDIO

32CL-2
32CL-4

COMPRESSOR/LIMITER
OWNER'S MANUAL

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IMPORTANT!

*** Read Before Using ***

CAUTION: The following *must* be observed to prevent malfunctioning and/or possible equipment damage.

- Before plugging the unit into the main AC line, make sure that all of the equipment following the compressor/limiter output lines is turned off or all of the inputs are turned down.
- The unit should be plugged in only when it has been established that the main AC line is supplying the correct voltage and frequency. US models are set up for 110 VAC at 60 Hz.
- Keep the unit away from excessive moisture.
- Allow only authorized technicians (consult your dealer) to open the unit. TDM Audio assumes no liability for damage or injuries.

Table of Contents

Introduction

Thank you for purchasing the TDM 32CL series compressor/limiter. These units are made from the finest components and engineered to exacting standards. Precision components are used in all critical circuitry for the finest sonic quality and performance. To get the most out of your new TDM compressor/limiter, please take a few minutes to review this manual and familiarize yourself with the proper operation of the unit.

The remainder of this section provides background information about the theory of compressor and limiter operation. Individuals that do not need this information may skip directly to the next section: *Mounting the Unit in a Rack.*

Compressor Fundamentals

Compressors are used to reduce the *dynamic range* of a sound signal. A signal's dynamic range is the difference between the loudest part of the signal and the quietest part. In theory, compressing the dynamic range of a signal results in the quietest parts getting a little louder and the loudest parts getting a little quieter. The way most compressors actually work, however, is by reducing the volume of the loudest parts. If desired, the overall volume of the output of the compressor may then be raised to bring the quietest parts up in volume without the loud parts being too loud.

The main use of a compressor is to even out the volume of a signal. For example, in a radio station, sometimes the announcer speaks softly while other times he or she might shout. This presents a problem because there is a limited modulation range that is acceptable for broadcasting. Radio stations use compressors so the signal is strong enough when the announcer is speaking quietly without causing overmodulation during the loud parts. The quality of the radio announcer's voice changes, but the actual volume of the signal is relatively constant. Even without the modulation limits, it is desirable while you are listening to the radio in your car to be able to hear the announcer over the road noise without being blown away when they get excited.

Compressors are used extensively in broadcasting, music recording, film-making, live sound reinforcement, and just about anywhere else sound equipment is found. In order to understand and operate a compressor, you need to be familiar with a few terms.

The first term you need to understand is *compression ratio.* The compression ratio is the mathematical expression of how much compression is taking place. For example, for a 2:1 compression ration (read "two to one"), the dynamic range is halved. The first number represents the dynamic range of the sound before compressing, and the second represents the dynamic range after compressing. Two to one means that if the dynamic range were two before compressing it would be one after compressing. A more realistic example would be a symphonic program which might have a dynamic range of around 70 dB before compressing. This same program would have a dynamic range of 35 dB after compressing at 2:1. If you flip the numbers over and place a line between them you get the fraction of the original dynamic range that describes the result. For example the result of 2:1 compression is ½ the original dynamic range. Although this is the theory, the reality is a little more complicated, so read on.

The problem with achieving a pure compression ratio comes from the way compressors work. Without knowing what the loudest and softest parts of the program are going to be, how can a compressor adjust its gain to achieve an exact 2:1 compression ratio? Well, the answer is that it can't. The way compressors work in the real world is with a *threshold.* The next term to familiarize yourself with is *threshold.*

The threshold is your compressor's way of letting you tell it what the softest part of the program will be. With this knowledge, the compressor can accomplish its goal. Take the 2:1 example again. A compressor with a 2:1 compression ratio reduces the gain by 1 dB for every 2 dB the input signal is over the threshold. If the threshold is -60 dBm, and the input signal is 20 dBm, which is 80 dB above the threshold, gain reduction will be 40 dB so the resulting output signal is -20 dBm. -20 dBm is 40 dB above the threshold so the dynamic range of the input is reduced by half. If you were to set the output gain to be exactly ½ of your expected maximum gain reduction (in this case ¼ of your expected input dynamic range), you would achieve true 2:1 compression. In such a setup, the loudest sound (at +20 dBm) would output 0 dBm, and the softest sound (at -60 dBm) would output -40 dBm. From -40 to 0 is a 40 dB dynamic range compressed equally at either half of the spectrum. It rarely really works this way, though.

One problem with the above scenario is that it raises the noise floor by 20 dB which is usually unacceptable.

Another problem is that such a scheme usually doesn't sound the way you would want it to. Generally, there is a range of sounds near the bottom end of the dynamic range of the program that you want to leave uncompressed in order to let the quietest parts of the program decay naturally without a lot of "breathing." *Breathing* describes the sound of the compressor bringing up the gain as things get quieter. You can hear the program noise increase, and the program decays unnaturally.

Another problem is that if the sound source is a microphone being reproduced through a loudspeaker, low thresholds cause feedback problems. The problem is that the microphone gain is much hotter when there is no sound than when there is. When a person speaking into such a microphone pauses, the compressor does the equivalent of cranking the volume of the microphone way up. This often results in feedback.

Low thresholds also cause unnatural sounding attacks. Suppose the threshold is at -60 dBm and the compressor is set to a 2:1 compression ratio. If the nominal level is 0 dBm, the signal gain is 30 dB hotter when there is no signal than when the nominal level is present. When the signal first hits, the initial output is really loud, and then you can hear the gain being reduced by the compressor. This is because it takes time to reduce the gain. In an ideal compressor, this gain reduction would happen instantaneously, but in the real world it never does.

The solution to these problems is to raise the threshold. In most setups, the compressor doesn't actually start compressing until the level is much higher than -60 dBm. This leaves a large portion of the dynamic range uncompressed, eliminates breathing, reduces output noise, provides for natural sounding attacks, and just generally works better. What it also means, though, is that the ratio no longer really represents the actual compression of the dynamic range. If the threshold is set at -20 dBm, and the loudest signal is +20 dBm, then only the portion of the range from -20 dBm to +20 dBm is compressed. For 2:1 compression, the gain reduction at +20 dB is 20 dB. If the dynamic range of the input signal ranges from -50 dBm to +20 dBm, the range is compressed from 70 to 50 dB. This is really only a 7:5 ratio, and compression only happens in the high 40 dB of the 70 dB range. Although this fact might be mathematically offensive, it results in much better sound quality.

If all of this math confuses you, you are not alone. You don't need to thoroughly understand it to work with compressors, though. It is presented so that you have the background necessary to understand how compressors work. Normally, you set the compression ratio based on how severe you want the compression (or gain reduction) to be, and you set the threshold to where you want the compression to begin. Ratios are usually set to a ballpark value based on the kind of sounds being compressed, and thresholds are generally fine-tuned to achieve exactly the desired affect.

Limiter Fundamentals

A limiter is really just a compressor with a really high compression ratio. One of the main real-world differences between a limiter and a compressor is that a limiter is a *hard-knee* device. The *knee* here is the knee of the curve that graphs input level vs. output level. Compressors tend to be *soft-knee* devices. A compressor eases across the threshold gently and then begins compressing the signal at the specified ratio. A limiter hits the threshold like a brick wall and then refuses to let the level go any higher. Limiters are used to set an absolute maximum level and then guarantee that the level will not pass it. A limiter has a ratio nearing infinity to one.

While compressors are often used to enhance sound quality, limiters are used mostly for protection purposes. A radio station might use a nice 4:1 compression ratio to even out the sound of the announcer's voice, but a limiter might be set to make sure that no matter what happens, the level doesn't go over the maximum modulation level allowed by law. Here the limiter is a protection mechanism. Limiters are used in large concert sound systems to prevent damage to speakers (and ears).

Limiters are much easier to use than compressors. It is not necessary to understand how the output level, compression ratio, and threshold interact to produce a certain effect. A limiter is used by simply setting the absolute maximum level as the threshold.

Mounting the Unit in a Rack

TDM 32CL series compressor/limiters can be mounted in any standard 19” rack. Each TDM 32CL series compressor/limiter takes up one rack space. To make mounting easier, lay the rack on its back with the equipment front panels facing up. Remove any rack screws from the part of the rack where you are planning to mount the compressor/limiter. Position the TDM 32CL series compressor/limiter in the rack as desired. Make sure the mounting holes in the compressor/limiter line up with the screw holes in the rack rails. Use four standard 10-32 rack screws for each compressor/limiter. We recommend that the rack screws have plastic washers to prevent damage to the paint on the face of the compressor/limiter. Install each screw loosely through a mounting hole in the compressor/limiter and into the rack. Do not tighten the screws until they are all in place. After all four screws are installed loosely, make sure the compressor/limiter is placed exactly as you desire and then tighten the four screws until they are nice and snug, but not overly tight.

Using the Security Cover

The 32CL series compressor/limiter can be ordered with an optional security cover. The security cover will only fit on units shipped for use with it. Units shipped without the security cover have knobs and buttons that protrude from the face of the unit for easy access and operation. Units shipped for use with a security cover have knobs and buttons that are recessed into the face of the unit so that the security cover can be installed.

The security cover is used to prevent unwanted tweaking of the compressor/limiter settings. If your compressor/limiter will be installed permanently for one particular purpose, and will be set up once to operate properly, use the security cover. An example of this kind of installation would be a compressor limiter installed for a public address/paging system. The unit would be set up so that the volume of speech coming through the system would be sufficient to hear and understand over ambient noise levels, but not too loud. Perhaps a professional contractor would set up the system correctly using special measurement equipment to guarantee compliance with local laws. Once this system was set up, you would not want anyone besides the contractor to change it. Security covers are recommended for cases like these.

Do not use the security cover if your unit will be used for a lot of different purposes, or will require adjustment often. For example, if you are installing your compressor/limiter in a recording studio, the security cover is a bad idea.

If you purchased your unit with a security cover, make sure all of your settings are correct, then install the cover. There are two holes in the face plate of the compressor/limiter—one on either end. Position the cover so that its two screws line up with the two holes. Then, using an Allen wrench, tighten the screws so that they are snug, but not overly tight.

If you purchased your unit without a security cover, but you wish to be able to use one, the unit can be converted. Contact your vendor, or call TDM (see *Contacting TDM* at the end of this manual) for information on how you can do this.

Hooking Up the Compressor/Limiter

Once your TDM compressor/limiter is installed in the rack, you are ready to hook it up to your equipment. Of course, the method used to hook up any compressor or limiter depends a great deal on how it will be used. We will try to give you the basics in this manual, but you may need to tailor the methods described here to your particular application.

What You'll Need

To connect your TDM 32CL series compressor/limiter to your equipment, you will need the following.

- **Power:** A power outlet should be located close enough to the unit so that you can plug it in. The TDM 32CL series compressor/limiters require a grounded (3-prong) outlet. If an outlet is not close enough, an extension cord or power strip may be used. Make sure you check the power rating on the extension cord or power strip to make sure that it exceeds the power requirements of all units plugged into it combined. TDM units have their power requirements marked on the rear of the unit.
- **Signal Cables:** Each channel to be connected requires two signal cables. One cable transmits the signal from your signal source to the compressor/limiter. The other transmits the signal from the compressor/limiter to the next item in the signal chain. This might be an amplifier, a mixing device, or more signal processing equipment.

Making Adapter Cables

In order to hook up your compressor/limiter to the rest of your equipment, it may be necessary to use special adapter cables. TDM 32CL series compressor/limiters can be shipped from the factory with either ¼" phone jacks, or with XLR connectors. In either case, inputs and outputs are balanced. The ¼" phone jacks are tip-ring-sleeve type for balancing. It is common to need adapters to go from ¼" to XLR or vice versa. It is very important that if you build your own adapters, you adhere to certain rules to make sure your TDM unit functions correctly.

If you have a TDM compressor/limiter with XLR connectors and you need to connect it to some other piece of equipment that has ¼" connectors, you might need to build XLR to ¼" adapters. If the other piece of equipment is balanced with tip-ring-sleeve jacks, then do the following.

- Wire pin 1 of the XLR connector to the sleeve of the tip-ring-sleeve plug.
- Wire pin 2 of the XLR connector to the tip of the tip-ring-sleeve plug.
- Wire pin 3 of the XLR connector to the ring of the tip-ring-sleeve plug.

If the other piece of equipment is not balanced, but has plain ¼" connectors, then do the following.

- Wire pins 1 and 3 both to the sleeve of the ¼" phone plug.
- Wire pin 2 to the tip of the phone plug.

It is very important that both pins 1 and 3 are wired to the sleeve. Some adapters leave pin 3 floating, and with TDM compressor/limiter units this will cause problems.

If you have a TDM 32CL series compressor/limiter with ¼” connectors, you may connect directly to other equipment using ¼” connectors whether the other equipment is balanced or not. If you wish to connect a TDM 32CL series compressor/limiter with ¼” connectors to another piece of equipment that has XLR connectors, and you are making the adapters yourself, use tip-ring-sleeve type ¼” phone plugs, and wire them this way.

- Wire pin 1 of the XLR connector to the sleeve of the tip-ring-sleeve plug.
- Wire pin 2 of the XLR connector to the tip of the tip-ring-sleeve plug.
- Wire pin 3 of the XLR connector to the ring of the tip-ring-sleeve plug.

The Basic Hook-Up

Begin by connecting the unit to a power source. Turn the power switch off. The power switch is located near the power cord at the rear of the unit. It is off when it is in the “out” position. Pressing the switch toggles it between off and on states. Next, plug the power cord that emerges from the back of the unit into an electrical outlet capable of supplying the correct voltage, current, and frequency. This information is printed on the rear panel of your TDM 32CL series compressor/limiter. Leave the unit turned off until all connections are made and you are ready to operate the unit.

The normal way to connect a compressor/limiter is to insert it into the signal path. A signal source is fed into the input of a channel, and the compressed and/or limited signal comes out of the output. For example, if you wanted to compress some music before it went out over a PA system, you would feed the music source into the input of a channel of the compressor/limiter (2 channels for stereo), and then feed the compressed signal from the output of the compressor/limiter into the amplifier.

Compressing and/or Limiting a Mix

If you are using a mixing console to mix several sound sources, and you would like to compress and/or limit the entire mix, use the following hookups.

- Connect the output from the mixer to the input of a channel of the compressor. If the mix is stereo, use two channels and connect the left mixer output to the input of one and the right mixer output to the input of the other. If there will be other processing in the signal path between the output of the mixer and the input of the amps or tape recorder or transmitter, make sure the compressor/limiter is the last in the chain.
- Connect the output of the TDM 32CL series compressor/limiter to the input of whatever you are sending the mix to. If you are mixing a recording, the output goes to the tape machine. If you are mixing a live event, the output goes to the sound system. If you are mixing for broadcast, the output goes to the transmitter. The idea is that you break the signal path between the mix and its destination and insert the TDM compressor/limiter into it. Again, if more than one signal processor is inserted, the TDM compressor/limiter should be the last thing in the chain. For example, if you are using your TDM compressor/limiter in a live

sound application where you have a mixing board, a master house equalizer, and the TDM compressor/limiter, connect the output of the mixer to the input of the equalizer. Then connect the output of the equalizer to the input of the TDM compressor/limiter. Finally, connect the output of the TDM compressor/limiter to the input of the sound reproducing system (either the amplifiers or the crossover in a multi-amped system). If the mix is stereo, use the two outputs of the channels that you connected the left and right inputs to. These outputs should feed whatever the left and right mix outputs would be feeding if you were not using the compressor/limiter.

Using Channel or Subgroup Inserts

Many mixing consoles provide channel inserts that let you insert signal processing equipment into the signal path of a single sound source. Some provide subgroup inserts so you can insert signal processing equipment into the signal path of a group of sound sources. This can be very useful. For example, when mixing a musical group, it is often desirable to create a subgroup for vocals, and then compress just that group. That way the dynamic range of all vocals can be evened out and kept in front of the mix without the level of the rest of the group affecting it. You don't want the vocals going into compression when the drummer hits the snare or it will be very difficult to understand them. Of course, it is also common to compress a single vocalist or instrument. In fact, it's quite common to see a subgroup for all background vocals with a single compressor, and the lead vocalist with his or her own compressor in concert situations.

There are two basic varieties of console inserts. One kind includes two ¼" phone jacks—one send and one return—that let you insert something into the signal path. With nothing plugged into the return, the two are internally connected together so the signal just passes through. You can also use just the send by itself as a channel output. Plugging anything into the return breaks the internal connection so that whatever is plugged into the return is what gets heard in the mix. To use this kind of mixer insert with the TDM 32CL series compressor/limiter, connect the send to the input of a channel of the compressor/limiter. Then connect the return to the output of the same channel of the compressor/limiter. You need one compressor/limiter channel for each mixer channel or subgroup being processed.

The other kind of console insert is actually more common. There is a single tip-ring-sleeve type connector for each channel or subgroup, and the send and return share a common ground lead. The usual configuration is for the ring to be the send and the tip to be the return. That way you can plug a ¼" phone plug halfway into the insert jack and use it as a channel output without breaking the signal path. Plugging it all the way in sets up an insert and breaks the internal signal path in the channel. To use this kind of channel insert with the TDM 32CL series compressor/limiter, you need to make a special insert cable for each channel that you want to insert a channel of compression into. To make an insert cable, use the following hookups.

- Make two cables, each with a ¼" phone plug on one end.
- On the other end, connect the ground (shield) leads of the two cables together.
- Solder the joined shield leads to the sleeve of a tip-ring-sleeve phone plug.
- Solder the hot lead of one of the two cables to the tip of the tip-ring-sleeve phone plug. Mark the plug on the other end of this cable as "Tip".

- Solder the hot lead of the other cable to the ring of the tip-ring-sleeve phone plug.

Now you have what resembles a “Y” cable, but one end is a tip-ring-sleeve plug, and the two ends that it splits out into are regular phone plugs. On the mixing console, plug the tip-ring-sleeve plug into the insert connector for the channel that you wish to insert the compressor into. Plug the ¼” phone plug marked “Tip” into the *output* of a channel of the compressor/limiter. Plug the remaining ¼” phone plug into the input of the same channel. If this setup does not work correctly, reverse the input and output because a few mixing consoles are set up to work the other way around.

A helpful hint: If your TDM compressor/limiter will mainly be used to insert into channels on a mixing console, you might want to create all of the insert cables and hook them up to the compressor/limiter in the rack. Mark the tip-ring-sleeve insert plug of each cable with the channel number that it corresponds to in the compressor/limiter. That way, if you want to put a compressor/limiter on a specific channel, you can just find the plug with the correct number, plug it into the channel insert, and adjust the corresponding channel on the compressor/limiter.

Compressing Electronic Instruments

The TDM 32CL series compressor/limiter can also be hooked up to directly compress electronic instruments. When electronic instruments are compressed, it is usually to produce a certain kind of sound or special effect rather than to smooth out the dynamic range. Connecting the unit to an electronic instrument rig is straightforward.

For best results, the signal that you feed into the TDM 32CL series compressor/limiter should be a line-level signal. Guitars and some keyboard instruments produce a signal that is much lower than a line-level signal. These low-level signals do not work very well with the TDM 32CL series compressor/limiters because often they are not hot enough to pass even the lowest thresholds, and because they result in noisy operation. Additionally, magnetic pickups like those found on passive electric guitars require special input impedance characteristics to sound right. That does not mean that you can’t use your TDM 32CL series compressor/limiter with electric guitars and other low-signal-level musical instruments. It simply means that the level of the signal must be amplified before it is fed into the compressor/limiter.

If your rig has an effects loop, you might want to insert the compressor/limiter into it. Effects loops are designed to operate at line level. To insert the TDM 32CL series compressor/limiter into an instrument effects loop, connect the effects loop send to a compressor/limiter channel’s input. Then connect the output of the same channel to the effects loop return.

With most effects loops this will work fine. Some amplifiers and instrument preamps have effects loops that do not feed 100% of the signal through the loop. These kinds of preamps and amplifiers will not work as well with a compressor/limiter. If you are using a separate preamp and amplifier, you can insert the unit between the preamp and amplifier in the signal chain. To do this, connect the output of the preamp to the input of a channel of the compressor/limiter. Connect the output of the same channel to the input of the amplifier.

Operating the Compressor/Limiter

Once you have mounted your TDM 32CL series compressor/limiter in a rack and connected all of the cables, the unit is ready for operation. At this time, turn all equipment on. It is always best to turn the equipment on in the order of the signal path from input to output and to turn it off in exactly the reverse order. For example, in a live sound setup you would turn on the mixing console, then any effects devices, then the equalizers, then the compressor/limiter, then the crossover, and finally the power amps. Turn everything off in the reverse order beginning with the power amps and ending up back at the mixing console.

It is not absolutely critical that you use this ordering, but it *is* absolutely critical for reinforcement systems that the power amplifiers are the last to be turned on and the first to be turned off. If the power amps are on when any of the other equipment is turned on or off, a loud pop through the speakers can result. This pop can very easily damage speakers. This is especially true in multi-amped systems, and with high-frequency drivers.

Front Panel Controls

At this time, familiarize yourself with the controls on the front panel of the unit. The 32CL-4 has four channels while the 32CL-2 has only two. All channels are identical, so you only need to understand how a single channel works. Here is a list of the controls for a single channel.

- **Output Level Control:** This control varies the output signal gain of the channel. Turning this knob clockwise raises the output gain. Counter-clockwise lowers the output gain. The output gain can be varied from -20 dB to +20 dB. Use this knob to adjust the level of the signal after the desired amount of compression has been achieved. In other words, set the threshold and ratio to achieve the desired gain reduction, then use the output level control to set the output signal to the desired level.
- **Compressor Threshold Control:** This knob lets you set the compressor threshold. The threshold is the level at which compression begins. For more information about thresholds, see *Compressor Fundamentals* in the beginning of this manual. The threshold can be varied from -40 dBm to +20 dBm.
- **Compressor Ratio Control:** This knob lets you set the compression ratio. The compression ratio determines the severity of gain reduction relative to how far the input level is over the threshold. For more information about compression ratios, see *Compressor Fundamentals.* The compression ratio can be varied from 1:1 (no compression) to 40:1 (very severe).
- **Compressor Attack Rate Button:** This button lets you select one of two pre-programmed attack rates for the compressor. The attack rate determines how quickly gain is reduced as the input level rises. Pressing the button in results in a fast attack rate. Leaving the button out results in a slow attack rate.
- **Compressor Release Rate Button:** This button lets you select one of two pre-programmed release rates for the compressor. The release rate determines how quickly gain is increased as the input level falls. Pressing the button in results in a fast release rate. Leaving the button out results in a slow release rate.

- **Limiter Threshold:** This knob sets the threshold of limiting. The threshold of limiting is the absolute maximum level that you ever want at the output of the compressor/limiter. For more information about limiter thresholds, see *Limiter Fundamentals.* The limiter threshold can be varied from -40 dBm to +20 dBm, and is an absolute maximum—it is not affected by the output level control. This means that the limiter threshold stays the same as the output level is varied. Another way to look at this is that the limiter threshold is an *output* threshold. Limiting begins when the *output* level reaches the threshold. The compressor threshold is an input threshold so compression begins when the input reaches the threshold.
- **Signal Light:** This light comes on when signal is present at the input of the channel.
- **Peak Light:** This light comes on when the output level reaches its peak. If the output signal is greater than the peak, audible distortion can result. It is okay if this light occasionally flashes, but if it flashes a lot or stays on consistently, reduce the output level by turning down the output level control, and find a way to get more gain out of some other component in the system that is *after* the compressor/limiter. For example, if you are compressing a live mix you might want to turn down the output level of the compressor/limiter and turn up the gain on the power amplifiers or on the crossover.
- **Compression Light:** This light comes on when the input level is over the compressor threshold. It indicates that compression is taking place.
- **Limit Light:** This light comes on when the output level is over the limiter threshold. It indicates that limiting is taking place.

Basic Compressing

This section will give inexperienced users the basic information they need to use the compressor. If you are new to compressors, expect a lot of trial and error to find the best settings for your particular needs.

The objective when compressing a signal is to get its dynamic range to the desired place. Most often, signals that need compression are just signals whose level varies too much and you want to reduce the dynamic range to something usable. Another way to look at this is that a signal needs compressing if when you turn it up so you can hear the quietest parts, the loudest parts are too loud.

Start with the output level at 0 (straight up), the threshold at 20 (fully clockwise), and the ratio at 1:1. This setting just passes the signal through with no compression. Start with the attack and release both in (fast), and make sure the limiter threshold is at 20 (fully clockwise) for no limiting.

Begin with some idea of the kind of compression ratio you need. For more information on compression ratios, see *Compressor Fundamentals.* Remember that you will generally not compress the quiet parts of the signal very much, only the louder parts. If your compression ratio is too low, your threshold will need to be set too low to achieve the correct amount of gain reduction. This will result in breathing and other undesirable side effects. If your compression ratio is too high, only the very loudest parts of the signal will be compressed, and they will be compressed severely. This will sound unnatural. It takes time and experimentation to fully understand all of the trade-offs involved so an intelligent choice can be made. The basic rule of thumb is that the less compression you expect to need, the lower the ratio you should use. Use the lowest possible ratio and the highest possible threshold to accomplish your goal. This will result in the most natural sound. When used properly, you should not be able to hear when the compressor is working.

The range of compression ratios commonly used varies from about 4:1 to about 20:1. Ratios outside this range are useful for achieving special effects, but they rarely sound natural. For speech, background music, and general program material, somewhere between 4:1 and 8:1 should be about right. If you are inexperienced, try closer to 4:1 first. If you notice that you have to set the threshold too low (more on setting the threshold later), raise the ratio up to about 6:1 and then raise the threshold and see if that works. After some time you should get to the point where you have a pretty good idea of what compression ratio you need for a particular application.

After selecting a compression ratio, adjust the output level so that the quietest parts of the signal are the right volume. Then slowly lower the threshold until the loudest parts of the signal are reduced in volume so that they, too, are the right volume. You can see when the signal passes the threshold and compression begins because the compression light will come on. If you have to lower the threshold a really long way after you first see the compression light come on, and you notice that the sound is unnatural, try moving the threshold back up, raising the compression ratio a little bit (not too much), and then lowering the threshold again. Usually, you will end up with a threshold that is not much lower than about -10 dBm. If it is a lot lower than this, you might need a higher compression ratio. Generally, the Compression light should only come on during the louder parts of the program. If it is staying on constantly, try using a higher ratio and a higher threshold.

Fast attack and release times are sufficient for most applications. A slow release time is sometimes used for speech or program material. A slow attack time is generally used to solve problems associated with having a slow release time.

Basic Limiting

Limiting is not very difficult. It’s a matter of finding the absolute maximum allowable level and setting the limiter’s threshold to that level. Generally, this is done by raising the signal to an unacceptably high level, then lowing the limiter threshold until the signal level is as high as is permissible, but no higher. For example, if you wanted to use the limiter to prevent clipping in an amplifier, you would raise the input signal until the amplifier began to clip, then lower the threshold of the limiter until the clipping stopped. You should then see the limit light on the limiter flashing on the same program peaks that originally caused clipping.

Compressing Speech

When compressing speech, it is of utmost importance that the resulting sound is natural and intelligible. A compression ratio of about 4:1 is sufficient for most speakers.

One problem that is common when compressing speech for multiple speakers is that the quietest speakers need a lot of gain to be heard, so the loudest speakers are constantly being compressed. This can become very noticeable because with a loud voice, every time the speaker pauses for a breath, the compressor releases the gain reduction so that the first syllable of the next phrase is unnaturally loud. Then you hear the compressor reduce the gain to a reasonable level again. This can become very annoying. The basic problem is that you generally want only the loudest parts of a program to be compressed, but when a boisterous speaker takes the microphone, the entire time he or she is speaking, he or she is in compression. The common way to deal with this problem is by using a slow release rate.

Slow release rates are used for applications where the input level remains fairly constant for a long period of time at one level, and then remains fairly constant for a long period of time at another level. You need compression to even out the levels, but you don't want the compressor wildly jumping around during a particular level. With a slow release rate, the compressor takes its time letting the gain go back up when the speaker pauses. If the speaker starts right back in again, the gain has not been increased significantly, so the sound is more natural.

One of the problems that can be caused by a slow release rate is the phenomenon of percussive sounds "punching holes in the program." If you are using a slow release rate, and the speaker makes a particularly percussive "P" sound into the microphone, the resulting transient causes the gain to be reduced substantially, and the gain takes quite a while to get back up to its previously correct level. The effect is that of the "P" punching a hole in the program. The solution to this problem is to use a slow attack rate. A slow attack rate causes the compressor to take more time to reduce the gain in response to a loud sound. The sound has to be loud for a reasonable length of time in order for the compressor to respond by reducing the gain. The result can be a more natural sound, but it is only useful when the sound level stays fairly constant for long periods of time. Slow attack and release rates form a sort of "gain inertia" where the gain has a healthy and natural resistance to change. For some applications, such as speech, this can be very useful.

Compressing Vocals

Compressing a singer is similar to compressing a speaker, except that the dynamic range is generally much greater, and the input level is constantly changing. You would usually not want to use slow attack and release rates with singers because the compressor would not react quickly enough to be of much use. For singers with a lot of dynamic range, you can use compression ratios of up to about 8:1 without sounding too unnatural. It is important that the threshold is not set too low because the dynamic nuances of the quiet parts of the sound can be lost. One problem that setting thresholds too low on vocalists can cause is an unnatural emphasis on sibilance and on breaths. These problems are common even with tame compressor settings sometimes, so it is often advisable to use side chains to deal with them. For more information on how to do this, see the next section *Using Side Chains.*

When compressing microphones used in live reinforcement, it is very important that the threshold is not set too low. Thresholds that are set too low can cause feedback problems. If you are compressing a vocalist, and you can't seem to get enough gain during the loud parts of the music, do not just keep pushing the fader up. If you do, when the music stops the microphone will probably immediately begin to feed back. Instead, realize that the loudest needed signal has changed from what you originally thought it was, and raise the compressor threshold until you get enough volume. Moving faders up and down constantly defeats the main purpose of compression—to manage the dynamic range of the vocalist so you don't have to.

Compressing Musical Instruments

Compressing musical instruments falls into two general categories. There is compressing musical instruments to manage their dynamic range, and there is compressing musical instruments to achieve a particular “sound” or effect.

Compressing musical instruments to manage their dynamic range is similar to compressing vocals. The main difference is that you can generally get away with more severe compression ratios on instruments without them sounding unnatural. Ratios of up to about 15:1 are common for instruments. It is common to put compressors on brass instruments, acoustic guitars, electric bass guitars, and many other instruments—even drums! Even though you can get away with more severe compression ratios with instruments, it is wise to use the lowest ratios and the highest thresholds unless you are going for a particular “sound.”

Overcompression has an unnatural sound, but for some instruments it can create a pleasing effect. The most common uses for this kind of compression in modern music are guitars, electric bass’s and acoustic drums (especially kick drums). The difference between this kind of compression and “normal” compression is that you use really severe compression ratios (up to about 20:1) and lower thresholds. When you are compressing to create an effect you want the unit to be in compression the whole time the instrumentalist is playing. This technique often requires that special precautions are taken to avoid excessive noise and feedback problems. These precautions include gating and special signal processing.

Compressing Program Material

Compressing program material is similar to compressing speech. In many cases, program material does not vary drastically in level from moment to moment. That makes it a good candidate for slow attack and/or release times, a low compression ratio, and a somewhat lower threshold than many other types of compression. These settings have the effect of evening out the overall level of the program over a long period of time without sounding too drastic or unnatural.

Using Side Chains

How Side Chains Work

To understand the concept of side chains, you must first understand the basics of compressor operation. The signal that you feed into a compressor channel is split and goes in two different directions. On the one side of this split feeds a level detector circuit that senses how loud the signal is. The other side goes through a voltage-controlled amplifier (VCA). A VCA is an amplifier circuit whose gain is determined by a control voltage. When the input level is below the threshold, the VCA just passes the signal through normally. When the input level rises above the threshold, the level detecting circuitry determines how far above the threshold the level is, and feeds a signal to the VCA to adjust its gain accordingly. The important concept here is that there are two signal paths, the one that gets compressed and output, and the one used to determine the level of the input signal.

Side chains are paths by which the level-detecting part of the signal is processed *before* the level detecting circuitry. For example, sometimes the signal is equalized before the level detector. Here is a picture illustrating this kind of side chain processing.

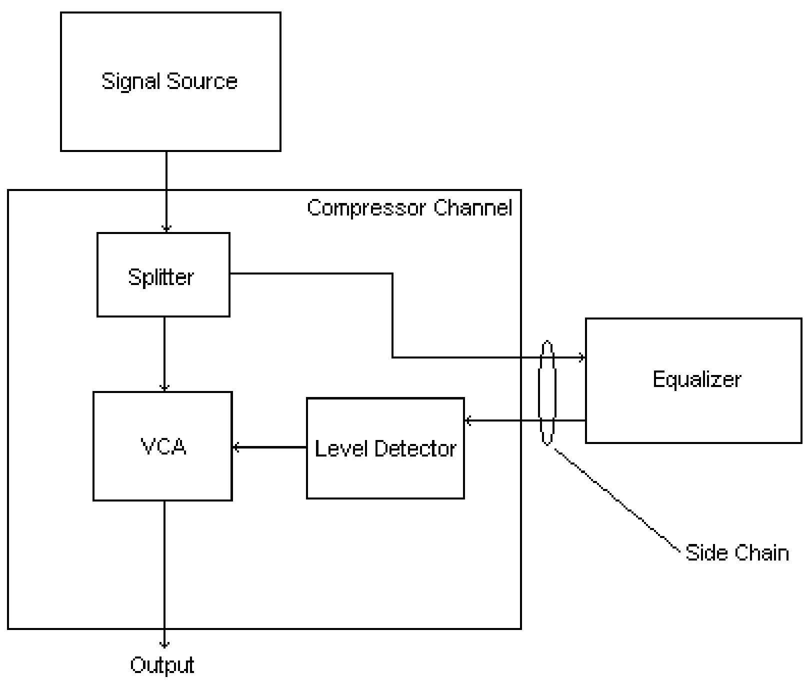

Figure 1 - Side Chain Equalization

As you can see in Figure 1, the path from the signal to the output does not pass through the equalizer. The equalizer only changes the signal that the level detector uses. This results in a change in the way the signal is compressed, but the signal itself is not altered by the side chain processing.

Uses for Side-Chain Compression

The possible uses for side chain compression are limited only by you imagination, and they are diverse enough to have entire books written about them. Often side chains are used in recording studios to create strange and interesting effects. Nonetheless, there are a few common uses for side chain compression that bear mentioning.

- **De-Emphasizing a Frequency Range:** If the side chain is fed through an equalizer that boosts a range of frequencies, the result is that the gain is reduced more when the input signal contains these frequencies than when other frequencies are present. In other words, the overall level of the output signal tends to be quieter when a certain range of frequencies is dominant. The equalizer does not affect the balance of frequencies of present in the signal at any one instant. Instead, the compressor reduces the gain of the signal when a certain range of frequencies dominates the input signal.

  One very practical use for this kind of compression is sibilance control. If the side chain equalizer boosts the high frequencies in the signal while reducing all other frequencies, then the gain of the overall signal is reduced when an overabundance of high frequencies is present. This keeps sibilance under control without reducing the natural high frequency content of the signal under normal circumstances. It lets you set a threshold, and reduces the overall level of the signal only when the high-frequency content is too hot. There is a special kind of device called a *de-esser* that is designed to do this job, and good de-essers are actually more sophisticated than compressors using side-chain equalization, but a compressor with a side chain will do the same job quite well under most circumstances.

  When compressing a vocalist, it is often desirable to insert some side chain equalization with low and low-mid frequencies rolled off and higher frequencies boosted. This can help reduce the undesirable emphasis that compression puts on sibilance and breaths.

  There are cases when other frequencies need to be de-emphasized naturally, and for those cases, side-chain compression is often the best choice. For example, side-chain compression is sometimes used on an electric bass to de-emphasize the upper midrange frequencies. When the bass player is plucking the strings with his or her fingers, the level of upper-midrange frequencies is under the threshold and the sound of the instrument is naturally reproduced. When the bass player is slapping or popping the strings, the compressor makes sure the upper-midrange attack is not overbearing without reducing the level of the actual tone.

  Using side chains to de-emphasize a frequency range is usually done with fast attack and release times.

- **Emphasizing a Frequency Range:** If the side chain is fed through an equalizer that cuts a range of frequencies, and the threshold is set so that the signal is in compression a good part of the time, the result is that the range of frequencies that is cut on the side chain equalizer is emphasized in the output. In other words, the gain of the signal is reduced by the compressor most of the time, but when the input signal is rich in the range of frequencies that is cut on the equalizer, the compressor does not reduce the gain as much. The result is that the output signal tends to be louder when certain frequencies are present. Again, the equalizer does not directly affect the balance of frequencies in the output signal. Rather, the compressor increases the overall gain of the output signal when certain frequencies are dominant.

An example of a common use for this kind of compression is low frequency emphasis in dance clubs. A compressor is often used to make sure that the sound level does not damage the hearing of club patrons or burn out the high frequency drivers in the sound system, but low frequencies are generally not as damaging to ears as high frequencies, and many dance clubs have a lot of low frequency power in the sound system, so they don't want the thundering low frequency content of dance mixes to drive the sound system into compression constantly. For such an application, a side chain is the natural choice. Low frequencies are reduced in the side chain equalizer, and the result is that the compressor is less likely to reduce the output gain due to an abundance of low frequency content in the signal. If you use a compressor with a side chain for this kind of protection, however, be aware that you are not providing any protection in the band of frequencies reduced on the side chain equalizer. You would be wise to set the limiter threshold to something lower than the absolute maximum that your sound system (or your listeners) can take.

- **Automatic Dimming of Program Material:** Suppose you have a paging system that plays music when nobody is speaking on the microphone. Instead of background music, however, this is foreground music, and it is fairly loud. When somebody needs to speak on the microphone, you don't want the music to just stop while the speaking is taking place because that would sound too unnatural. Instead, you would like it to just get quieter so that the person speaking can be heard, and then to go back up in volume automatically when the speaker finishes.

  This is a fairly common scenario, and the solution is side-chain compression. This kind of side chain compression is unusual, however, because the program material itself is not actually processed in the side chain. The side chain *send* is simply ignored, and a signal from the microphone preamplifier or mixer is fed into the side chain *return.* With this setup, the level of the microphone signal determines the amount of gain reduction. The louder the signal at the microphone, the quieter the program material gets. The threshold is set fairly high so that ambient noise on the microphone does not reduce the level of the program material. The compression ratio, however, is typically quite high (20:1 or more) so the volume reduction is dramatic enough to permit the speaker to be heard. The attack time is set to fast so that the volume is reduced immediately when someone starts talking on the microphone. The release time, however, is set to slow so that the volume does not go back up when the person pauses to take a breath. When they are finished speaking, the volume of the program material gradually ramps back up to normal.

  This kind of side chain compression is common in clubs with DJs, and in radio and television broadcasting.

Connections for Side Chain Operation

Each channel has a side chain connector. This connector is a tip-ring-sleeve type ¼" phone plug. The side chain send and return share a common ground which is connected to the sleeve. The tip is the signal send for the side chain and the ring is the signal return.

To use a side chain, make an insert cable (see *Using Channel or Subgroup Inserts*), and plug its tip-ring-sleeve connector into the side chain connector on the back of the TDM 32CL series compressor/limiter. Plug the connector on the other end that corresponds with the tip into the input of your side processing chain. Plug the connector on the other end that corresponds to the ring into the output of your side processing chain.

TDM Option Cards

The TDM 32CL series compressor/limiters have internal connectors for option cards. A variety of different option cards is available from TDM. Plugging a card into the connector inserts the circuitry on the card into the side chain. The most common card to use for this application is the parametric equalizer card. By simply plugging in the card and tuning its frequency response, a 32CL series compressor/limiter can be set up to emphasize or de-emphasize a range of frequencies without the need for external equalizers and cables. In cases where the unit will be permanently set up a certain way, using option cards is much more cost effective than purchasing a dedicated external equalizer just for side chain processing. The cards are installed internally, so once they are set up, they are under the cover and their settings cannot easily be changed. This is an advantage in situations where you don't want the settings changed, but it is an obvious disadvantage in cases where the settings will change often. In these situations, an external equalizer is recommended. For information about what cards are available, contact your vendor or call TDM (see *Contacting TDM*).

Troubleshooting and Support

This section details various problems that you might encounter when using any piece of signal processing equipment, and the possible causes and solutions. It also tells how to contact TDM when you need service or support for your 32CL series compressor/limiter.

No Signal Output

Make sure that the unit is plugged in, turned on, and that the power light on the front panel is illuminated.

Look for the signal light on the front of the unit. If this light on, or is flashing in time with the input signal, then the unit is getting a signal from the source. If not, check the source to make sure it is working correctly. If the source seems to be working correctly, but the signal light is not coming on, check the source by bypassing the TDM 32CL series compressor/limiter. In other words, try plugging the signal source output into whatever the output of the compressor/limiter is plugged into. If you do this and there is still no signal, the problem is either that the signal source is not providing a signal, or that the equipment that should be reproducing the signal isn't functioning properly. If you bypass the compressor/limiter and everything works correctly, then the signal source is fine.

If the signal source is not providing a signal to the compressor/limiter, check the cabling between the source and the compressor/limiter. Try substituting another cable that is known to be good. If the source is a channel insert on a mixing console, make sure that the send and return are not reversed. If it is an insert with a tip-ring-sleeve connector, try reversing the send and return to see if perhaps you have it hooked up backwards. If all of this fails, it's likely that your signal source is malfunctioning.

If the signal source is providing a signal to the compressor/limiter, but you are still getting no output signal, check to make sure that the problem is not with the equipment that should be reproducing the signal. Again, you can try bypassing the compressor/limiter and plugging the known-good signal source directly into next piece of equipment in the chain after the compressor/limiter. If it works correctly when you do that, then you know that both the source and the reproduction equipment are working properly. In this case, check the cable that connects the output of the compressor/limiter to the next item in the chain. Try replacing this cable with one that is known to be good. If this does not fix the problem, the compressor/limiter might be malfunctioning. Contact your vendor, or call TDM for support and/or service (see *Contacting TDM*).

Distortion

To determine the cause of the distortion, try systematically removing each piece of signal processing equipment from the chain, one at a time. After a piece of equipment is removed from the chain (by plugging the piece of equipment before it directly into the piece of equipment after it), listen to the system and determine if the distortion is still present. When you remove a piece of equipment and the distortion goes away, then it is likely that this particular piece of equipment is the cause of the distortion. If none of the signal processing units in the chain is causing the distortion, then either it is present in the signal source, or there is a problem with your sound reproducing equipment.

If you determine that the TDM 32CL series compressor/limiter is the cause of a distortion problem, make sure that the unit is plugged into a proper power source. Read the back panel of the unit for the correct supply voltage and frequency (US models are set up for 110 VAC at 60 Hz). Using a unit designed for 220 volt operation with a 110 volt outlet can cause distortion.

Check the peak light for the channel that you suspect is distorting. If the peak light is on during the audible distortion then you are overdriving the compressor/limiter. Try reducing the output gain by turning down the gain knob on the channel. If this fixes the problem, you need to find a way to get more gain out of some piece of equipment further down the chain. If this does not fix the problem, try reducing the level of the signal that you are feeding into the compressor/limiter.

If the peak light is not lit, the unit is plugged into the correct power source, the signal feeding the unit is clean, and you are still getting distortion out of the output, contact your vendor, or call TDM for support and/or service (see *Contacting TDM*).

Excessive Noise

The TDM 32CL series compressor/limiter has an excellent signal to noise ratio. If you hear excessive noise in your system, try to determine its origin systematically. Remove each piece of processing gear from the signal chain one at a time until you hear the noise go away. If none of the signal processing units is the cause of the noise, then the noise is probably present in your signal source.

If you suspect that the TDM 32CL series compressor/limiter is the cause of your noise problem, make sure the unit is plugged into the correct power source. Read the rear panel of the unit to determine the correct voltage and frequency (US models are set up for 110 VAC at 60 Hz). Using a unit designed for 220 volt operation plugged into a 110 volt outlet can cause very noisy operation of the unit.

Check to make sure that the input signal light is on most of the time while an input signal is present. If the input signal light rarely or never comes on, then the input signal is probably not strong enough. Using any piece of signal processing equipment with an extremely weak input signal can cause noise problems. If possible, increase the level of your signal source so that the signal light is on most of the time, and the peak light flashes occasionally at the loudest points in the signal. Be careful not to increase it too far or distortion can result. Note that when you raise the level of the input signal, you must raise the threshold of the compressor channel and lower the output level of the compressor channel to compensate.

Make sure that your compressor threshold is not too low. A threshold that is too low can cause a lot of noise when no signal is present. If the compression light is on constantly while an input signal is present, and you hear a lot of noise when the input signal is silent, chances are this is your problem. Another symptom of this problem is having the output level of the channel turned up really high, or having the level of some other piece of equipment after the compressor/limiter really high. In these cases, when you remove the compressor/limiter from the chain, you usually get a signal that much too loud. That means that you are compressing the signal extremely hard, and that when no input is present, the unreasonably high gain in the system is producing a lot of noise.

If you check out all of these possible causes and you still can't resolve the problem, contact your vendor, or call TDM for support and/or service (see *Contacting TDM*).

60 Hertz Hum or Buzz

60 Hertz hum or buzz in a system can be extremely difficult to track down because it is usually not a problem with any one piece of equipment. It is usually caused by how the entire system is connected and grounded.

To fix a hum or buzz in a system, suspect any piece of equipment that gets a ground connection from more than one place. These problems are called "Ground Loops" and the technical explanation of why they cause problems is that there is actually a voltage difference between the two different grounds. The problem is most often caused by a single piece of equipment grounded to two different power sources that are located some distance apart. For example, a mixing console is plugged into a grounded outlet at the back of an auditorium, and the power amplifiers are plugged into a different outlet 100 feet away at the stage. The mixing console is connected by shielded cable to the amplifiers and the shield is grounded. This causes both the mixer and the amps to be individually grounded, and each gets another ground from the other through the shielded cable.

A problem like this can be fixed in several different ways. The mixer ground could be lifted. This is commonly done by plugging the mixer's three-prong plug into a two-prong grounding adapter (you can get these at any hardware store), and plugging that into the outlet. This effectively disconnects the mixer's ground lead from the outlet so that the mixer is now grounded only to the amplifiers. The ground could also be lifted at the amplifiers so that they are grounded only to the mixer. If the cable connecting the mixer to the amplifiers is a balanced (3-wire) type, the ground can be floated at either end of this cable by disconnecting the wire connected to pin 1 of the XLR adapter at one end or the other (but not both). Sometimes, because of the particular setup, you will have to try several of these options before finding one that works.

> ***CAUTION: Check local codes and regulations for rules pertaining to electrical grounding. It may be illegal in some places to lift the ground of a piece of equipment—especially if this piece of equipment is installed publicly.***

Another common cause of ground loops is direct input (or DI) boxes. These let you plug an instrument such as a guitar or bass directly into a microphone input. The problem is that the person playing the guitar or bass might be using an amplifier or some other signal processing equipment on the stage that is plugged into a grounded outlet. This creates a ground loop between their setup and the grounded outlet that the mixing console is plugged into. Fortunately, many DI boxes have a ground lift switch that you can use to break this ground loop. Because they are so convenient, ground lifts on DI boxes are often the first option tried when a hum or buzz surfaces.

If you suspect that there is a problem with your TDM 32CL series compressor/limiter that is causing a hum or buzz, try removing the unit from the system and plugging its output directly into a power amplifier with a speaker attached. Make sure that the TDM 32CL series compressor/limiter is plugged into the same electrical outlet as the amplifier. If the hum or buzz is still present, there might be a problem with the unit. In this case, contact your vendor, or call

TDM for support and/or service (see *Contacting TDM*). If the hum or buzz is not still present, the problem is somewhere else in the system, and is probably a ground loop.

Contacting TDM

If you have a problem with your TDM 32CL series compressor/limiter that you cannot solve using this troubleshooting guide, contact the vendor where you purchased the unit. If you need further assistance, you can call TDM at (818) 765-6200 during normal business hours (9 AM to 5:30 PM Pacific time). Our FAX number is (818) 765-8262. Our E-mail is support@tdmaudio.com and our Web site is TDMAUDIO.COM. Your satisfaction is our business, and we are happy to help you get the most out of your TDM 32CL series compressor/limiter.

Specifications

Frequency Response	+0-1.0 dB 10 Hz - 20K
Total Harmonic Distortion	
R_L > 2 k ohms	<0.009% THD
Maximum Output Level	
R_L > 2 k ohms	+22 dBu (6.2 volts) @ <.05% THD 20-20 kHz
Maximum Voltage Gain	+20 dB
Hum and Noise (20 Hz-20 kHz)	
Av = 0 dB	<-93 dBu
Signal-To-Noise Ratio	115 dB
Controls	
Input Control	-20 to +20 dB
Compressor Threshold	-40 to +20 dB
Compressor Ratio	1:1 to 40:1
Compressor Attack	
Switch	
Slow	10msec
Fast	10usec
Compressor Release	
Switch	
Slow	500msec
Fast	100msec
Limiter Threshold	-40 to +20 dB
LED Indicators	
Signal Present	-20 dB Green
Signal Peak	+10 dB Red
Compressor Active	Green
Limiter Active	Green
Power Indication	Green
Side Chain	
Phone Jack insert	
Sleeve	Ground
Ring	Sig. Out
Tip	Sig. In
Internal Insertion Connector	Accepts TDM Opt. Cards
Output Type	Floating and balanced
Connectors	XLR and 1/4 Phone
Output Impedance	300 Ohms
Input Type	Balanced and Differential
Connectors	XLR and 1/4 Phone
Input Impedance	20K Ohms
32CL-2/4	
Dimensions (W x H x D)	19 in. x 1.75 in. x 8 in.
Weight (boxed)	8 lbs.
0 dBu = 0.775 v rms