# Omnia® SST

Powerful, full featured audio processing software for your PC.

USER'S MANUAL v8.53

For part numbers: 3001-00070, 3001-00071, 3001-00085-000 and 3001-00086-000

Telos Alliance®

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User Warnings and Cautions

The installation and service instructions in this manual are for use by qualified personnel only. To avoid electric shock, do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so. Refer all servicing to qualified personnel

This instrument has an autoranging line voltage input. Ensure the power voltage is within the specified range of 100-240VAC. The ~ symbol, if used, indicates an alternating current supply.

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

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

CAUTION: HAZARDOUS VOLTAGES

The instrument power supply incorporates an internal fuse. Hazardous voltages may still be present on some of the primary parts even when the fuse has blown. If fuse replacement is required, replace fuse only with same type and value for continued protection against fire.

WARNING:

The product's power cord is the primary disconnect device. The socket outlet should be located near the device and easily accessible. The unit should not be located such that access to the power cord is impaired. If the unit is incorporated into an equipment rack, an easily accessible safety disconnect device should be included in the rack design.

To reduce the risk of electrical shock, do not expose this product to rain or moisture. This unit is for indoor use only.

This equipment requires the free flow of air for adequate cooling. Do not block the ventilation openings on the rear and sides of the unit. Failure to allow proper ventilation could damage the unit or create a fire hazard. Do not place the units on a carpet, bedding, or other materials that could interfere with any panel ventilation openings.

If the equipment is used in a manner not specified by the manufacturer, the protection provided by the equipment may be impaired.

USA CLASS A COMPUTING DEVICE INFORMATION TO USER.

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, subpart 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. Note: objectionable interference to TV or radio reception can occur if other devices are connected to this device without the use of shielded interconnect cables. FCC rules require the use of shielded cables.

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"This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the radio interference regulations of the Canadian department of communications."

"Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques (de Class A) prescrites dans le règlement sur le brouillage radioélectrique édicté par le ministère des communications du Canada."

CE CONFORMANCE INFORMATION:

This device complies with the requirements of the EEC council directives:

- ◆ 93/68/EEC (CE MARKING)
- ◆ 73/23/EEC (SAFETY – LOW VOLTAGE DIRECTIVE)
- ◆ 89/336/EEC (ELECTROMAGNETIC COMPATIBILITY)

Conformity is declared to those standards: EN50081-1, EN50082-1.

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Updates

Telos Omnia.9 features and operations are determined largely by software. The Telos Alliance strives to provide the most stable and feature-rich software available. We encourage you to check for software updates from time to time by visiting our website or by contacting us directly.

Feedback

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Non-emergency technical support is available at Support@TelosAlliance.com.

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The Telos Web site has a variety of information that may be useful for product selection and support. The URL is https://www.telosalliance.com/Telos .

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Congratulations on your new Telos Alliance product!

The gang here at Telos is committed to shaping the future of audio by delivering innovative, intuitive solutions that inspire our customers to create the most exciting and engaging audio experiences imaginable.

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Telos Alliance

Omnia SST User's Manual

v8.53 (February 2021)

Introduction

Thank you for purchasing Omnia SST, a fully-featured software-based audio processor for FM and digital broadcast. SST offers a powerful suite of tools to both repair and process audio.

This document should be used in conjunction with our Omnia SST Quick Start Guide (QSG). While the QSG gives you a general orientation to setting up SST, especially audio hardware configuration, these pages are intended as a reference guide, describing what various controls, meters and elements of SST do. Near the end of this manual, you will also find a guide to Omnia μMPX® (Micro MPX) Radio's First Composite FM Codec: High-quality MPX over a 320kbps IP connection.

As is the case with other PC software, it is assumed you have moderate to advanced (PC) computer knowledge, audio hardware configuration experience and networking skills. In the case of this very specialized audio software, it is also assumed that you have moderate to advanced familiarity with broadcast processing, audio signal chains, RDS, FM transmission, and other systems that may feed or be fed by a computer running Omnia SST.

But whether you are a pro or a newcomer to audio processing, the best approach to learn what SST can do is to simply install it on a computer, input source audio and start navigating various screens and trying different settings. Building your own mental model of how different controls and settings can make a difference to your sound will be the best teacher. We encourage you to experiment, try different settings, alter and make your own presets, and generally explore. If you get stuck or have questions, feel free to write us at:

support@telosalliance.com

Home Page

Input/Output (Same as "Soundcards" settings panel)

Sample Rate

Selects the sample rate to use for audio input.

Normally, the output sample rate will be identical to the input sample rate. The only exception is when you are not using ASIO, the sample rate is lower than 128 kHz, and FM output is used.

In this case, if you use sample rates above 48 kHz, the audio will be downsampled before processing and upsampled again afterwards. 88.2 and 176.4 kHz are downsampled to 44.1 kHz, 96 and 192 kHz are downsampled to 48 kHz.

The sample rate has a small effect on the latency and CPU load. At 48 kHz there is a bit more data that needs to be processed, which slightly increases the CPU load. The latency is a bit smaller because a block of the same number of samples is a bit smaller. If you use 32 kHz or a multiple thereof, the CPU load is a lot smaller - but you should only use this low a sample rate it if you don't need any audio output above approximately 14 kHz.

If you want to use the FM output with stereo and RDS encoding, and you are using ASIO, then the sample rate must be set to at least 128 kHz. Samples rates of 176.4 or 192 kHz are preferred. If you don't use ASIO for the output, the output sound card will automatically be opened with a high enough sample rate to send out the stereo and RDS signals.

Important:

> **If you are <u>not</u> using ASIO, you need to make sure that the Windows sound card driver is configured to use the same sample rate that you have selected in Omnia SST, or Windows will resample it. This causes audible artifacts and may cause issues with the FM output (or it may not work at all). Open the properties for the audio card you are using and click on the "Advanced" tab. Set the "Default Format" to a mode with a sample rate that matches the one selected in SST.**

ASIO

Audio Stream Input/Output (ASIO) is a computer sound card driver protocol for digital audio specified by Steinberg, providing a low-latency and high fidelity interface between a software application and a computer's sound card.   ASIO bypasses the normal audio path from a user application through layers of intermediary Windows operating system software so that an application connects directly to the sound card hardware. Each layer that is bypassed means a reduction in latency. ASIO offers a relatively simple way of accessing multiple audio inputs and outputs independently.

The ASIO control selects whether or not to use an ASIO device driver for your hardware. These drivers are generally more stable, offer better control over sound cards, and reduce delay between input and output. It is strongly suggested that ASIO be used if possible. The ASIO buffer size can be adjusted through the card's ASIO driver control panel. This functionality varies between cards. Consult your audio card documentation for more details.

A note about other driver models:

If an ASIO driver for you audio device is not available, SST supports Kernel Streaming, WASAPI, and MME driver models for accessing audio devices. If ASIO is not available, Kernel Streaming (WDM) is preferred, talks directly to the device driver, and will offer the best performance. Some devices may requires use of WASAPI (the "new" Windows standard for communicating with audio resources) or MME (the "older" Windows standard). Not all operating systems, hardware, or drivers support all methods. You may need to try each to determine which works best with your audio hardware.

ASIO (On/Off)
Turns ASIO use on or off. This is a global setting and will enable or disable the ability to use an ASIO device for inputs and outputs.

ASIO Device ID
Selects which ASIO driver and device to use for ASIO I/O selections. Only one device can be selected here, though 3rd party drivers such as ASIO4ALL allow additional flexibility in routing between multiple ASIO devices.

Reduce buffer clicking (On/Off)

This function protects against clicks and pops due to buffer underruns.

When the ASIO buffer size is set very small (to minimize latency, for example) in some cases a block of audio may not yet be available when it needs to be sent to the ASIO driver, resulting in audible clicks.

Enabling this setting lets Omnia SST anticipate the data that would be sent to the sound card and send that predicted data instead. This masks most of the clicks and pops, but occasional underruns may still occur.

Turning this setting temporarily can help you determine the minimum acceptable buffer size before clicks start to occur. Once you determine this threshold, increase the buffer size slightly and re-enabling it the setting to give an additional margin of protection once the buffer size is set appropriately.   For best results, keep this option enabled during normal (on-air) operation.

Main Input

Selects the audio inputs that will be used for the main processing path.

Sound Card Input

ASIO Input Left/Right

If ASIO is enabled, this selects which inputs to use on the selected ASIO device. These settings will override the "Input Device ID" setting, which is used to select a WDM device. Selecting "No ASIO" will disable ASIO for the input. In addition, ASIO Input Right is disabled when "No ASIO" is selected in "ASIO Input Left".

Input Device ID

Selects a WDM device to use for input. This setting is overridden when "ASIO Input Left" is set to anything other than "No ASIO".

Buffer Usage

Displays the input buffer status. Normally this bar will remain nearly full.

Input Level Correction

Input Gain

Adjusts the input level to SST. Input metering should reach around 0 dB on peaks.

Many facilities allow a lot of headroom in their signal path, with reference levels potentially as low as -24 dBFS for analog signals of +4 dBu. The factory presets in Omnia SST were designed for input levels around 0 dB on the input meters. With lower input levels than that, certain stages such as the declipper, noise removal, and AGC may not function as effectively as they can.

Of course, it is still possible to accommodate lower input signal levels with a large amount of headroom. With this slider you can adjust the level such that under normal circumstances the peaks are at about 0 dB. If peaks are occasionally louder, that's no problem. Unlike some environments, the signal does not clip at 0 on the input metering.

For best results, adjust the input gain of the sound card mixer <u>before</u> adjusting this control. Setting gain stages is important in any system, and adding gain at this stage can potentially increase any noise introduced by the sound card.

Balance

Adjusts for different input levels on left and right channel.

If the left and right channel input levels aren't perfectly equal (which might happen for example if you're using analog equipment), this slider allows you to correct it. A negative value means that the left channel is boosted, a positive value means that the right channel is boosted. A good way to calibrate balance is to input mono audio and adjust this slider until the input meter levels are visually equal for left and right.

Most other devices (including amplifiers) may reduce the level of a channel instead of increasing it. The reason to increase the level is that otherwise it's more difficult to show both the original and the modified (by Input gain and Balance) volume levels - one channel could be made louder while the other could be made softer.

Tilt Correction (Input)
Corrects tilt problems in the input signal.

A tilt control tilts the frequency content of the audio signal by simultaneously boosting the treble and cutting the bass frequencies, or vice versa.

If you have an analog signal path into Omnia SST, you might have some tilt issues. This means that a square wave does not look like a square wave at the input anymore—i.e. it has "tilt".

Input tilt correction can compensate for this, and slightly improves bass response. This also helps the declipper to function at its best.

While tilt correction on the input is useful, it is far more important on the output as it affects systems downstream.

Correction enabled (On/Off)

Enables or disables input tilt correction.

RC

RC value of the first high-pass filter.

Tilt is often caused by a Resistor and Capacitor (RC) circuit, which acts as a highpass filter. For example, a sound card might use an RC circuit to remove DC offset from a signal. Unfortunately, this also reduces the audio quality and makes processing more difficult.

The tilt correction filter in Omnia SST inverts the effect of an RC circuit, thus restoring the original signal (except for the DC offset).

This slider sets the RC value of the RC circuit to be inverted.

Pre- and Post-correction waveform displays

When enabled, the "Tilt Correction" panel includes two waveform displays, pre- and post-correction. Feed a square wave into the input and adjust the "RC" control until the waveform on the right is square.

Secondary Input

Selects a second audio input, which can be used for backup (failover) audio or SCA purposes (including external RDS).

Secondary Input

Global settings for the secondary audio input.

Input 2 On/Off

Enables or disables the secondary audio input.

ASIO Input 2 Left/Right

If ASIO is enabled, selects which inputs to use on the selected ASIO device. These settings will override the "Input 2 Device ID" setting, which is used to select a WDM device. Selecting "No ASIO" will disable ASIO for the input. In addition, "ASIO Input 2 Right" is disabled when "No ASIO" is selected in "ASIO Input 2 Left". If the secondary input is used for SCA, the left input will be SCA Channel 1 and the right input will be SCA Channel 2.

Input 2 Device ID

Selects a WDM device to use for the secondary audio input. This setting is overridden and disabled when "ASIO Input 2 Left" is set to anything other than "No ASIO".

Buffer Usage

Displays buffer status for input 2. Normally this bar should remain nearly full.

Usage

Selects how the secondary audio input will be used

Use as backup (On/Off)

Selects the secondary audio input as a backup audio source, which can either be switched to air based on silence, or continuously mixed with the main source.

Always add to main input (On/Off)

Continuously mixes the secondary audio input with the main input. This is useful, for example, if one input is used for a satellite program feed, and the second is used for local insertion. This feature is also useful for networks which have both "regional" and "local" programming fed to the transmitter site from different studio locations, one of which is silent while the other is active. This setting will be disabled if "Use as Backup" is "Off".

Use secondary input as external RDS input (On/Off)

Allows input of an external RDS generator signal. Internally, the pilot will be synchronized to the incoming RDS signal - any incoming pilot signal will be ignored. A poor quality or week RDS signal might result in problems, including the internal stereo pilot generator will take over. Those factors aside, the input level does not matter. This setting will be disabled if "Use as backup" is "On".

Threshold level/Minimum Silence

Specifies the silence threshold level and timeout before switching to the secondary audio input as backup (assuming "Use as Backup" is "On" and "Always add to main input" is "Off"). If "Use as Backup" is "Off" or "Always add to main input" is "On" these settings will be disabled.

Input Level Correction

Gain

Compensates for low signal levels on the secondary audio input. There is no audio level metering for the secondary audio input.

FM Output

These settings control the audio card output used for the FM MPX signal.

FM Output

FM Output (On/Off)

Turns the FM composite (MPX) output on or off.

ASIO Output Left/1 (Right/2)

If ASIO is enabled, selects which physical outputs to use on the selected ASIO device for the MPX outputs. These settings will override the "Device ID" setting, which is used to select a WDM device. Selecting "No ASIO" will disable ASIO for the output. In addition, "ASIO Output Right/2" is disabled when "No ASIO" is selected in "ASIO Output Left/1".

Device ID

Selects a WDM device to use for the MPX output. This setting is overridden and disabled when "ASIO Output Left/1" is set to anything other than "No ASIO".

Volume (MPX Level)

Controls the overall level of the MPX output. Normally this would be set such that peaks are well controlled and the pilot level is close to the setting configured in "FM Options > Stereo Generator > Pilot Volume" under the main configuration menu.

Buffer Size

Adjusts the amount of buffering or diversity delay to apply to the MPX output. Can be set up to 12.5 seconds.

Buffer Usage

Displays the status of the MPX output buffer. If this bar is not staying filled, audio dropouts will result. Increase the buffer size to ensure this buffer is staying filled.

FM Tilt Correction

As with the audio card inputs, many sound cards introduce "tilt" on their output due to output filtering and lack of DC coupling. This can cause overshoots and result in lower modulation (as much as 3-5% or more) as levels are reduced to account for the overshoot while keeping within appropriate modulation limits.

While ideally an audio card with DC coupled inputs and outputs would be used to avoid the issue, SST has tilt correction on both input and output to compensate for this.

Correction Enabled (On/Off)

Enables or disables tilt correction on the FM output.

RC

RC value of the first highpass filter.

Tilt is often caused by a Resistor and Capacitor (RC) circuit, which acts as a highpass filter. For example, a sound card might use an RC circuit to remove DC offset from a signal. Unfortunately, this also reduces the audio quality and makes processing more difficult.

The tilt correction filter in Omnia SST inverts the effect of an RC circuit, thus restoring the original signal (except for the DC offset).

This slider sets the RC value of the RC circuit to be inverted.

Pre/Post Correction Scopes

When enabled, the "Tilt Correction" panel includes two waveform displays, pre- and post-correction. After adjusting for input tilt, enable a square wave on the internal test signal generator (described below) then connect the card's output to its input. Adjust the RC slider until the right-hand waveform display is square.

Test Signal

Generates various test signals on the outputs for calibration purposes.

Generate Test Tone (On/Off)

Enables or disables the internal test signal generator and replaces the processed audio at the outputs with the selected test signal.

Available test signals include:

- ◆ Sine
- ◆ Smooth square
- ◆ Square
- ◆ 400 Hz Sine
- ◆ (various) Bessel Null
- ◆ Sawtooth w/pause (for AM)
- ◆ Sawtooth flat bottom (for AM)

Frequency

Adjusts the output frequency of the selected tone. This control is applicable for sine, square, or smooth square and is ignored for other tone selections with fixed frequencies.

Single Frequency Networks

Omnia SST supports applying a sample level delay on an MPX output. This is useful for single frequency network or booster applications where precision alignment between signals is required.

Delay one of the MPX outputs (On/Off)

Enables or disables the MPX output delay.

Channel (MPX Out 1/MPX Out 2)

Selects the MPX output channel to delay.

Invert

Inverts polarity on the selected MPX output channel.

Delay

For fine-tuning Adjusts the amount of delay (measured in samples) applied to the selected channel from 0 to 192 samples.

Web/DAB+/HD Output

This section contains controls for a secondary audio processing output, independent of the FM MPX and low latency outputs. It is typically used for a web stream, DAB/HD, or processed monitoring where latency is not a concern. The signal patch point for this output is selectable.

Sound Card Output

Normal Output (On/Off)

Turns the secondary processing output on or off.

ASIO Output Left/Right

If ASIO is enabled, selects which outputs to use on the selected ASIO device for the secondary processing path. These settings will override the "Device ID" setting, which is used to select a WDM device. Selecting "No ASIO" will disable ASIO for the output. In addition, "ASIO Output Right" is disabled when "No ASIO" is selected in "ASIO Output Left".

Device ID

Selects a WDM device to use for the secondary audio output. This setting is overridden and disabled when "ASIO Output Left" is set to anything other than "No ASIO".

Web/DAB+/HD Output

Controls the output level of the secondary processing output.

Buffer Size/Diversity Delay

Adjusts the amount of buffering or diversity delay to apply to the secondary audio processing output.

Buffer usage

Displays the status of the secondary processing path output buffer. If this bar is not staying filled, audio dropouts will result. Increase the buffer size to ensure this buffer is staying filled.

Signal Selection
drop down menu:

De-Emphasized version of FM output

Follows the FM processing path and emulates what listeners might hear on an FM radio after pre-emphasis and de-emphasis.

Separately processed streaming/HD/DAB+ output

Provides a processed output separate from the FM processing path.

Separately processed pre-emphasized left/right FM output

Pre-emphasized stereo FM output without composite clipper.

Separately processed de-emphasized left/right FM output

De-emphasized stereo FM output without composite clipper

Input without processing

Duplicates the input signal without applying processing. Useful for creating an additional unprocessed output for monitoring or other purposes.

Low Latency Monitor Output

These settings control the low latency output for studio monitoring. This output does not include clipping.

Sound Card Output

Low Latency Output (On/Off)

Turns the low latency output on or off.

ASIO Output Left/Right

If ASIO is enabled, selects which physical outputs to use on the selected ASIO device for the low latency output. These settings will override the "Device ID" setting, which is used to select a WDM device. Selecting "No ASIO" will disable ASIO for the output. In addition, "ASIO Output Right" is disabled when "No ASIO" is selected in "ASIO Output Left".

Device ID

Selects a WDM device to use for the low latency output. This setting is overridden and disabled when "ASIO Output Left" is set to anything other than "No ASIO".

Note:

> **Using WDM rather than ASIO with the low latency output will add sufficient delay to make it unsuitable for low latency usage.**

Volume

Controls the output level of the low latency monitor output. Since there is no clipping on this output, the volume may need to be reduced to account for any spikes in the audio level.

Buffer Size

Adjusts the amount of buffering to apply to the low latency output. Set this as low as possible to minimize latency while still maintaining a stable output.

Buffer Usage

Displays the status of the low latency output buffer. If this bar is not staying filled, audio dropouts will result. Increase the buffer size to ensure this buffer is staying filled.

Clean Up

The "Clean Up" sections are used for pre-conditioning the audio prior to processing. This includes removing hum, de-clipping, restoration of lossy encoded audio, correcting for azimuth errors in recordings, noise reduction, and restoring dynamic range.

Each block on this page has an "On/Off" toggle, as well as a "Hear" control. These controls allow you to easily listen to the results of each block while fine-tuning the processing. "On/Off" will enable or bypass each block individually, while "Hear" allows you to hear the processing up to that point, bypassing all downstream blocks.

The "On/Off" controls are available inside each block as well. Some blocks also contain a "Delta" control which allows you to hear the effect of that just that block.

Note that toggling these controls affects ALL outputs and WILL be heard on the air. There is no "Preview" output available. The "Hear" and "Delta" controls can be disabled in the system settings to avoid inadvertent activation.

Dehummer

The "Dehummer" can be used to learn and remove any constant noise from a source. For constant noise at specific frequencies, the noise can usually be filtered entirely from the source audio. For other noise (such as hiss or other broadband noise) it is typically possible to achieve several dB of attenuation without significantly affecting the source audio.

Learn (On/Off)

Used to "learn" the noise signature to be removed. With only the noise source applied to the input, turn the "Learn" control "On" for several seconds. This samples the noise such that a filter can be constructed to remove it.

Reset Analysis Data

Once a particular noise signature has been "Learned" it becomes a single, fixed filter. If the input source changes over time, the noise signature will need to be updated as this filter may no longer apply.   This reset control will discard any learned analysis data in preparation for learning a new noise signature.

Declipper

Although FM processors typically perform clipping as a final processing stage, many modern recordings are already clipped before they even reach the listener. The declipper can account for this and repair clipped audio prior to processing.

Input is analog or resampled (On/Off)

Turn this toggle on if the input is analog, or if it may have been been resampled.

If the input is analog or resampled, the position in the audio where samples are taken 'moves', which makes it more difficult to distinguish between samples that are clipped and not clipped.

When this toggle is on, if a sample is detected as clipped, the 2 surrounding samples will also be marked as clipped. This does mean that the set of 'good' samples is reduced, which makes restoration a bit harder. But the opposite, not detecting a sample as clipped while it actually is clipped, is far less desirable.

Input can be compressed-MP3 etc. (On/Off)

Select this function if the incoming audio has been encoded to MP3 or other bit-reduced format at some point.

After lossy encoding such as MP3, audio gets spread in time. If a sample is clipped, the samples adjacent to it will also get a part of the distortion. As a result, samples close to clipped samples which have a relatively high value all need to be treated as clipped.

Maximum peak restoration

Sets the maximum size of the peaks that will be recreated by the declipper.

Certain waveforms (dense, block waveforms for example) would cause a nearly infinite peak level. This slider forces the recreated peak level to stay below a certain point relative to the original level of the sample.

Setting this value too low may result in some remaining distortion after declipping.

DeLossifier

Lossy audio encoding (such as MP3) reduces the amount of data needed to store or stream audio by using psychoacoustic models to 'throw away' content that the codec deems 'nearly inaudible'. The trade-off of such data reduction schemes is that they can cause audio artifacts (especially at lower bitrates).

In addition to the fact that these artifacts are often already quite audible, the processing that is done by SST (or any other processor) may further accentuate such artifacts, countering the codec's assumptions of what is or is not audible.

The DeLossifier attempts to detect and clean up artifacts introduced due to lossy audio encoding before the audio hits the processing stage.

Pre-Ringing
*Adds considerable latency

One of the typical artifacts in MP3 and other lossy compressed files is pre-ringing. This means that a sound that should start suddenly has a sort of 'pre-echo'. Instead of starting suddenly you may first hear it come up softly, and then there's a sudden jump when the sound was really supposed to start. This gives the typical 'smeared' sound and loss of 'punch' on lower bitrate MP3s.

This filter attempts to detect pre-ringing, and reduce it.

Important: this filter uses a look-ahead mechanism in order to determine whether a sound is caused by pre-ringing or it is actually part of the signal. Look-ahead systems require buffering that can add considerable latency.

Kill pre-ringing (On/Off)

Turns the pre-ringing filter on or off.

Delta button

Outputs only the difference (delta) introduced by the pre-ringing filter, allowing you to hear its effect.

Spectral Hole Filler

Attempts to fill "holes" in the audio spectrum caused by lossy audio encoding. A "before" and "after" spectrogram are provided for reference/comparison.

Fill Spectral Holes (On/Off)

Turns the "Fill Spectral Holes" function on or off.

Delta button

Outputs only the difference (delta) added by the "Fill Spectral Holes" function, allowing you to hear its effect.

Azimuth

Attempts repair of azimuth (phase) errors.

Azimuth errors are often present in tape recordings, and also on poorly mastered CDs. Phase problems cause very unpleasant artifacts when playing a recording in mono or through a surround system. Even normal stereo playback may sometimes sound a bit "odd". Any large phase offset is automatically detected and removed by this filter.

This filter only works properly if audio on the left and right channels is similar. If this is not the case for a long period of time, the azimuth correction will slowly be reduced.

Image phase amplifier maximum angle

Sets the maximum amount of phase correction to be applied (up to 180 degrees)

Noise Removal

The noise gate removes background noise, which is mainly useful for older recordings from analog source material. Gentle use of this filter can also improve audio from some CDs by removing quantization noise.

Noise Removal

Global on/off control of how much noise is removed.

Total Noise Removal

Controls the maximum amount of noise to be removed.

Band Settings

Fine-tuning of the amount of noise to be removed per frequency band.

Relative band {1-7} noise removal

The maximum amount of noise to be removed per frequency band, relative to Total noise level.

Delta button

Outputs only the difference (delta) removed by the "Noise Removal" function, allowing you to hear its effect.

Natural Dynamics

In addition to being heavily clipped, modern recordings often lack any dynamic range. It's much easier for an audio processor to work with music that has some dynamic range rather than trying to further compress already heavily compressed recordings.

"Natural Dynamics" attempts to restore dynamic range and "punch" to heavily compressed recordings while not altering audio that already contains sufficient dynamic range.

Delta button

Outputs only the difference (delta) of the "Natural Dynamics" function allowing you to hear its effect.

Natural Dynamics

Main settings for the "Natural Dynamics" section

Strength

Increases or decreases the overall effect of "Natural Dynamics".

Bands

Overview of the band splits for the natural dynamics section. There are no adjustments in this section.

Quick Adjust

Contains a series of controls that manipulate key "processing" components described in the next section. Quick Adjust acts as a set of "VIP" controls and is useful for when you want to "tweak" a preset that has just been loaded, without going into all the depth and complexity of fuller control layers.   The "0" points of these controls will be whatever was loaded via the preset.

Loudness

Trims the amount of drive to the final clipper--this control has the biggest effect on overall loudness.

Texture

"Master" controls for the multiband section. These will impact how "busy" the audio sounds, trading openness for aggressive density.

Density
Adjusts master drive to the multiband sections.

Attack/Release speeds
Speeds up or slows down reaction time in the multiband sections.

Enhance (Lows, Mids, Presence, Highs)

Controls the "spectral balance" in the multiband section by groups of bands (depending on how many are loaded with the preset).

For a preset with 5 bands:

Lows = Band 1

Mids = Band 2

Presence = Band 3 and 4

Highs = Band 5

For a preset with 7 bands:

Lows = Bands 1 and 2

Mids = Bands 3 and 4

Presence = Bands 5 and 6

Highs = Band 7

Density adjusts drive to each of the respective band(s), while gain adjusts final output gain. If enabled, the "Bass In Your Face" control adjusts the amount of bass harmonic enhancement.

Stereo Enhance

Adjusts the amount of stereo enhancement effect applied to the audio.

L-R enhancement
Controls the maximum amount of boost applied to the L-R component of the input signal.

Strength
Adjusts the overall amount of Multiband Stereo Boost. Higher values sound 'wider', but extreme settings may increase the chance of multipath distortion.

Processing

The "Processing" section of SST is where all controls for the main audio processing blocks are grouped. This includes phase rotation, AGC, stereo enhancement, EQ, bass and high frequency enhancement, multiband processing, bandpass filtering, single band compression, and the final clipper.

Phase

Phase Rotation makes the peaks of asymmetrical waveforms more symmetrical. In other words, if a peak has a high amount of energy in the positive direction and very little in the negative direction, energy will be added in the negative direction. The same is true of peaks in the negative direction—energy will be added in the positive direction

This protects the compressors, limiters and clipper against certain types of sounds that can be difficult to process and tend to cause distortion. Examples are voices (especially female voices) and instruments like trumpets. These types of sounds are very asymmetrical with large peaks in one direction. If there is also lot of low frequency energy present (such as a song with heavy bass) it can cause additional issues with these waveforms due to intermodulation distortion.

Phase Rotation
Global settings for phase rotation.

Phase Rotation (On/Off)

Enables or disables phase rotation.

Sound Shaping
Additional controls for phase processing, including adding frequency specific phase delay.

Phase Delay (On/Off)

Enables or disables frequency specific delay adjustment, intentionally adding nonlinearities to the processing.

Delay versus frequency graph adjustment

Allows phase to be adjusted at specific frequencies. This can be useful to make more "boomy" bass sounds (for example), by emulating the nonlinearities found in certain analog processors. The horizontal axis of the graph is frequency, and the vertical axis is the amount of additional phase delay to be applied at that frequency. Curves can be modified by drawing them directly on the graph with the cursor.

Auto EQ

*Will cause increased CPU usage when enabled

This filter is a dynamic re-equalizer that operates before the AGC in the processing path. Using the familiar Bands user interface found in MB1 and MB2, it can be configured to precisely match bands already in use elsewhere. Subject to user settings, this filter will attempt to keep levels in the bands at a Target level with respect to each other. Spectrum Correction is not a compressor and does not affect actual levels, overall. Furthermore, using it does not require any subsequent filters to be altered to accommodate it. If this filter is boosting the level in some band(s), it is reducing some other band(s).

Without Auto EQ, the multiband sections are responsible for two jobs: 1) Controlling audio levels, and 2) Dynamically re-equalizing incoming audio. This is usually not a problem, but it can become one on some material.

Let's say incoming audio usually produces about 10dB of gain reduction in MB1 and all bands are roughly equal in the amount of gain reduction they're doing. Now a song comes along that has very little high end. The top band of the multiband section will release and could end up doing almost no work at all. That is suboptimal, because the multiband now has no margin of error. If the amount of highs drops even further, there's nothing the multiband can do to correct it, and the audio will become muddy and dull. That's where Auto EQ comes in, to fix material that is imbalanced before it hits any further filtering.

When audio is playing that is properly balanced, per the settings, this filter will have virtually no effect on the audio. When doing large amounts of correction, the filter should be unobtrusive. It simply fixes the audio relatively slowly while attempting to be as transparent as possible.

The user interface for this filter resembles the multiband compressor UI, but has some important distinctions. When the filter is doing no work, you'll see nothing in the interface (no colored bars). Bars stretching to the right, in shades of blue indicate a boost in level. Bars toward the left, in shades of red, indicate a cut to the volume in that band.

The individual controls are discussed below.

Auto EQ (Main Controls)

Adjust time

The speed at which bands can move up and down in relation to the incoming audio. Unlike a compressor, running Spectrum Correction at a faster speed does not make audio more dense. It simply corrects spectral imbalances more quickly. Extensive listening and watching will assist in figuring out the best speed.

Relative kick speedup

After the filter has increased level in a band, a loud sound might kick in which causes the filter to need to reduce it again. Sometimes, that correction might seem too abrupt. This control allows that downward correction to be slowed down and potentially sound more natural. Lower values correspond to slower downward correction.

Band Controls

Effect

Adjusts the effect of the Auto EQ in each band

Effect strength
This filter does not have to fully correct spectral imbalances. Using this slider will make the filter do less correction. If band 1 is 6dB too quiet, and this is set to 50%, only 3dB of correction will happen.

Minimum level
The maximum amount of reduction the filter can make for this band.

Maximum level
The maximum amount of increase the filter is allowed to provide for this band.

Levels

Threshold
Sets the relative point at which the Auto EQ will begin to act in a given band.

Channel linking
It is possible that the left/right balance could be incorrect even within a single band. Use this control to specify how much the correction should be allowed to vary between the channels.

Band mix
After correction has been applied, it is possible to still increase or reduce the output of the band.

Bands

Displays the crossover points between bands. This display does not allow modifying the crossover points.

AGC

AGC (Automatic Gain Control) attempts to make the audio level constant by gradually increasing and decreasing the gain. If sudden level jumps are present in the audio, it can remove these transients without lowering the output level too much. This avoids pumping and similar annoying effects.

Leveler

Controls the overall behavior of the AGC section.

Drive

Adds additional gain to the AGC input to compensate for low input levels or to provide greater opportunity for low level audio to increase.

Target Level

Sets the target maximum output level from the AGC. This control essentially serves as the "drive" to the following stages of processing, including multiband.

Ratio

Determines the change in output level as compared to the change in input level. Lower ratios deliver a more open sound while higher settings provide increased density and tighter level control.

Window

Sets a window in dB in which the ACG will not increase or decrease gain, providing a "dead band" during which no processing occurs in this section. Larger values will provide increased dynamics and a more open, less "busy" sound but will also require downstream sections of the processor to work harder.

Gate Slowdown

Slows the AGC activity when the input level drops below this value to prevent rush up of noise or other artifacts. This also prevents large swings in levels between pauses in audio.

Side Chain

An EQ in the sidechain of the AGC to control the frequencies at which it responds. This can help prevent "pumping" of the AGC with heavy bass or midrange frequencies, for example.

Graph/EQ

Allows "drawing" an EQ curve for the sidechain, and displays an RTA of the incoming audio.

Transient Protection

Prevents radical level changes due to sudden increases in incoming audio levels by temporarily overriding the normally slow-acting AGC with a faster-acting AGC.

Threshold

If the volume suddenly increases by more than this amount, AGC transient protection is activated.

If you set this level too low, AGC transient protection will be activated too soon and there will be a loss of dynamics. If set too high, overly loud audio will be let through to the next processing steps. When transient protection is active, a portion of the AGC meter will change to a different color.

Slope

Determines the range between the threshold amount, and maximum amount of transient protection.

Power Bass

Adds additional low end punch to tracks that don't contain much bass energy—particularly certain older recordings. This results in a much deeper and warmer sound for both old and new music. On tracks that already contain a considerable amount of bass, there is little added effect.

Power Bass (On/Off)

Enables or disables the Power Bass section.

Lowpass

Determines the frequency below which bass will be added.

Maximum Boost

Controls the maximum amount of boost applied to bass frequencies.

Threshold

Adjusts the detection threshold of the Power Bass section. If the detected amount of existing bass content is above this threshold, additional bass will not be added.

Power Highs

Similar in function to Power Bass, but for high frequencies. This results in a much brighter and consistent sound for tracks lacking high frequency energy.

Power Highs (On/Off)

Enables or disables the Power Highs section.

Highpass

Determines the frequency above which high frequencies will be added.

Maximum Boost

Controls the maximum amount of boost applied to high frequencies.

Threshold

Adjusts the detection threshold of the Power Highs section. If the detected amount of existing high frequency content is above this threshold, additional high frequency energy will not be added.

Compressor

This is an additional wideband compressor following the Leveler, Power Bass, and Power Highs

Compressor

Turns the compressor stage of the AGC on or off.

Relative threshold level

Controls the input level above which the compressor becomes active, relative to the AGC target output level setting in the leveler section

Window

Sets a window in dB in which the ACG will not increase or decrease gain, providing a "dead band" during which no processing occurs in this section. Larger values will provide increased dynamics and a more open, less "busy" sound but will also require downstream sections of the processor to work harder.

Ratio

Determines the change in output level as compared to the change in input level. Lower ratios deliver a more open sound while higher settings provide increased density and tighter level control.

Attack

Controls how quickly the compressor will respond to changes in audio level above the threshold. Smaller numbers will result in a faster attack time. Move the control to the left to slow down attack and to the right to speed up attack.

Release

This control determines how quickly the gain reduction value recovers when the audio level falls below the threshold.

Gate freeze

"Freezes" the action of the compressor gain reduction when audio falls below this value.

Stereo

Tools to enhance the stereo separation.

Delta button

Outputs only the difference (delta) of the "Stereo" function to hear its effect.

Wideband delayed stereo enhancer

This feature adds a copy of the L-R (stereo, difference between channels) information to the original stereo signal which creates a sparkling stereo effect. In cases when the stereo information in the input signal is already very heavy, or almost non-existent, the enhancer switches off automatically. The effect is more pronounced on transients and can in some cases enhance bass response.

Wideband delayed stereo enhancer (On/Off)

Turns the wideband delayed stereo function on or off. This will grey out the associated controls.

L-R enhancement

Controls the maximum amount of boost applied to the L-R component of the input signal.

L-R Delay

Adjusts the amount of delay of the L-R component that is added to the original stereo signal. When using more delay, the sound appears to 'open up', but extreme settings can start to sound unnatural.

When set to 0, there is no added delay.

Threshold

Controls the level below which no additional L-R information will be added. This value is relative to the L+R level.

Multiband stereo enhancer

This section uses a multiband enhancer to increase the level of L-R information.

Multiband Stereo Enhancer (On/Off)

Enables or disables the Multiband Stereo Boost section.

Strength

Adjusts the overall amount of Multiband Stereo Boost. Higher values sound 'wider', but extreme settings may increase the chance of multipath distortion.

Equalizer

The equalizer section is fairly self explanatory. It affects the frequency response of the signal prior to the multiband compressors and subsequent stages. The interface allows "drawing" the desired EQ curve, and displays an RTA showing the input signal.

Parametric EQ

Allows use of standard parametric EQ controls rather than "drawing" the desired curve. There are 3 bands of fully parametric EQ, and two "shelving" EQ (Bass, Highs).

The resulting curve will be displayed in the RTA window. When enabled, any changes previously made by the "drawing" method are discarded.

Bass & Highs

This section deals with adding harmonic energy to both low and high frequencies.

Bass In Your Face 1 & 2

Bass In Your Face generates harmonics which appear to be missing from the original signal. It is designed to generate bass at a specific frequency which complements the rest of the signal and sounds natural.

Delta buttons

Each Bass In Your Face section has a "Delta" button. Toggle this button to hear the effect of ONLY that section.

Bass In Your Face 1 & 2 (On/Off)

These switches will enable or disable each Bass In Your Face section individually. The associated controls will be greyed out when the corresponding section is disabled.

Peak Frequency

Controls the center frequency around which each Bass In Your Face section will generate additional energy.

Strength

The relative "intensity" of the additional generated bass energy. Values close to 1.0 will give the most "natural" sound.

Phat Bass

Generates additional bass boost with harmonic energy.

Delta button

Allows you to hear the effect of ONLY the "Bass harmonics" section.

Phat Bass (On/Off)

Enables or disables the "Phat Bass" section. The additional controls in this section will be greyed out if this switch is turned off.

Strength

Determines the amount of bass harmonic energy that will be added

Up to

Determines the frequency up to which the maximum amount of bass boost will occur. Set the frequency higher to include more mid-bass enhancement. Excessively high settings can cause distortion on some material.

Slope to

No bass boost will occur at frequencies above this setting. Absolute Highs

Adds high frequency harmonic content.

Absolute Highs

Delta button

Toggle this button to hear the effect of ONLY the "Absolute highs" section.

Absolute Highs (On/Off)

Enables or disables the "Absolute highs" section. The remaining controls in this section will be greyed out if this switch is turned off.

Generate highs above

Sets the frequency above which additional high frequency harmonic energy will be generated.

Multiband 1 & 2

Main

Density

Adds additional gain to all bands to compensate for low input levels or to provide greater opportunity for low level audio to increase like a typical "Drive" control, but automatically applies an equal amount of makeup gain to the output. Higher density levels will result in more activity in the multiband limiters.

Master Attack/Master Release

Simultaneously affects the attack and release times set in the band speed controls. Sliding the control to the left results in slower times while moving them to the right results in faster times.

Band coupling

Controls the degree to which bands track one another. A setting of 0% allows the bands to act completely independently. Higher settings force all bands to track more closely resulting in more of a wideband effect.

Master Gate

Gate Freeze

Stops or "freezes" the release action of the multiband compressors when audio falls below the selected value.

Gate Slowdown

Slows down the release action of the multiband compressors when audio falls below this value. This is normally set at a higher level than the Gate Freeze control to create a window in which the release time slows down but doesn't stop completely.

Band Controls

G/R Meters

These meters indicate the amount of gain reduction in each band and mirror the multiband gain reduction meters elsewhere in the interface. The crossover frequency for each band is also indicated here.

"Hear" (Band solo)

This control allows you to listen to the action of each band individually and is helpful when fine-tuning a specific band.

Speed

This page contains individual attack and release speed controls for each band.

Attack
Controls how quickly each band responds to changes in audio level above the threshold. Smaller numbers will result in a faster attack time. Move the control to the left to slow down attack and to the right to speed up attack.

Release
This control determines how quickly the gain reduction value recovers when the audio level falls below the threshold.

Levels

This page contains controls that affect input and ouput levels in each band.

Density
Operates in the same way as the Main Density control, but for individual bands.

Threshold
Controls the input level above which the compressor becomes active.

Limit
Adjusts the drive to the multiband limiters. Moving the sliders to the right increases limiter activity and density.

Band Mix
Adjusts the final output level of each band. In Multiband 1, adjusting these controls will in turn affect the amount drive to Multiband 2. In Multiband 2, these controls will affect the amount of drive to the final clipper. Bear in mind that since Multiband 2 is the final processing stage before clipping, extreme adjustments here can cause audible artifacts from the clipper.

Sound

Controls that affect the gain reduction behavior of each band in the multiband section.

Feedback
Adjusts the balance between feed back and feed forward operation. In feed back mode (100%), the output level is used for detection. In feed forward (0%), the input level is used.

Ratio
Determines the change in output level as compared to the change in input level for individual bands. Lower ratios deliver a more open sound while higher settings provide increased density and tighter level control.

Knee

Adjusts the "curve" of the compression knee at the threshold. Higher values will make this transition softer, lower values will make it more abrupt.

Window

Sets a window in dB in which each bacn of the compressor will not increase or decrease gain, providing a "dead band" during which no processing occurs in this section. Larger values will provide increased dynamics and a more open, less "busy" sound but will also require down-stream sections of the processor to work harder.

Detection

Determines whether the compressor will use a peak-based or RMS-based algorithm.

Peak mode will tend to react to a single small spike in the sound. RMS mode responds more like human hearing does, but may cause pumping with low frequencies.

Gate

The "gate" controls will prevent the multiband compressor gain reduction in each band from moving back toward zero during periods of low or no audio.

Gate Freeze

Freezes the release action of each individual band when audio falls below this value.

Gate Slowdown

Slows down the release action of each individual band when audio falls below this value. This is normally set at a higher level than the Gate Freeze control to create a window in which the release time slows down but doesn't stop completely.

Bands

This section will display the band crossovers, as well as an RTA of the input and output signal from the multiband compressor. The grey portion of the bars represents the input signal, and the yellow portion the output.

Bandpass

The bandpass filter consists of highpass and lowpass filters, will allow only frequencies above or below a certain point to pass to the processor output.

Highpass

Passes only frequencies above a certain point.

Delta button

Toggle this button to hear the effect of ONLY the "Highpass" filter.

Highpass frequency

Adjusts the frequency above which audio will pass.

Lowpass

Passes only frequencies below a certain point.

Delta button

Toggle this button to hear the effect of ONLY the "Lowpass" filter.

Lowpass frequency

Adjusts the frequency below which audio will pass.

Singleband

An additional single band compressor which follows the previous processing stages, prior to the final clipper.

Delta button

Toggle this button to hear the effect of ONLY the single band compressor.

Speed

This group of controls affect the reaction time of the single band compressor.

Attack

Controls how quickly the compressor will respond to changes in audio level above the threshold. Smaller numbers will result in a faster attack time. Move the control to the left to slow down attack and to the right to speed up attack.

Release

This control determines how quickly the gain reduction value recovers when the audio level falls below the threshold.

Window

Sets a window in dB in which the ACG will not increase or decrease gain, providing a "dead band" during which no processing occurs in this section. Larger values will provide increased dynamics and a more open, less "busy" sound but will also require downstream sections of the processor to work harder.

Levels

The controls in this panel will affect the overall input and output levels of the single band compressor.

Density

Adds additional gain to compensate for low input levels or to provide greater opportunity for low level audio to increase like a typical "Drive" control, but automatically applies an equal amount of makeup gain to the output. Higher density levels will result in more activity in the single band compressor.

Threshold

Controls the input level above which the compressor becomes active.

Limiter

Adjusts the drive to a limiter integrated with this single band compressor. Moving the slider to the right increases limiter activity and density.

Sound

These two settings control the overall behavior of this single band compressor section.

Feedback

Adjusts the balance between feed back and feed forward operation. In feed back mode (100%), the output level is used for detection. In feed forward (0%), the input level is used.

Ratio

Determines the change in output level as compared to the change in input level. Lower ratios deliver a more open sound while higher settings provide increased density and tighter level control.

Gate

The "gate" controls will prevent the singleband compressor gain reduction value from moving back toward zero during periods of low or no audio.

Gate

Freezes the release action of the singleband compressor when audio falls below this value.

Gate Slowdown

Slows down the release action of the singleband compressor when audio falls below this value. This is normally set at a higher level than the Gate Freeze control to create a window in which the release time slows down but doesn't stop completely.

Clipper

The clipper is the final stage of processing, and does a significant amount of "heavy lifting" in Omnia SST prior to output. This stage is where adjustments with the biggest impact on both distortion and loudness are found. Intermodulation distortion is minimized as much as possible. The result is audio that can be extremely loud, yet still quite dynamic with very little distortion.

Clipper
Main controls for the clipper section

Clipper Drive (Loudness)

This control adjusts the input level to the final clipper and will have the largest effect on overall loudness.

Relative Web/HD/Dab Clipper Drive

The Web/HD/DAB output has the ability to use a separate clipper path when its output mode is set to "Separately Processed". This control adjusts the level of that path, relative to the main clipper path.

Sound
This panel contains controls that affect the overall texture of the clipper.

Sparking Highs (On/Off)

Enabling the "Sparkling Highs" control preserves the openness and enhances the texture of very high frequencies, especially when the incoming program material has a large degree of high frequency content.

Dirty Voices (Phat Mids)

This feature warms up and enhances mid-range frequencies, particularly male and lower female vocal ranges.

Bass Shape
The controls in the "Bass Shape" panel help set the "texture" of the bass.

Strength

Unlike most bass clippers, the one in Omnia SST can (but does not have to) create a square wave when processing strong low frequency content. Setting the "Strength" control at "0" (0%) will always prevent square waving the bass resulting in a very smooth low end texture, but one that does not offer the "slam" of a traditional Omnia clipper even on the initial peak of a bass transient waveform.

Allow harmonics up to

This control sets the frequency of the highest harmonic produced by the bass clipper. Higher settings will produce more and higher harmonics and create a more complex and frequency-rich bass sound.

Distortion

These controls further affect the texture and feel of the bass clipper, and will have the biggest effects on loudness versus distortion.

Delta button

Toggle this button to hear the effect of ONLY this stage. This will allow you to hear the distortion being cancelled in the bass clipper.

Intermodulation distortion protection (On/Off)

This control will defeat IMD protection in the clipper. Unless you specifically want the presence of intermodulation distortion (where strong low frequency material can attenuate or audibly distort higher frequencies) we strongly recommend the "Intermodulation Distortion Protection" control be left enabled.

Clip threshold

The "Clip Threshold" control is the point at which bass clipping will occur under normal circumstances and is analogous to a traditional "bass clipper threshold" control.

Protection threshold

When bass at the level set by the "Clip Threshold" control would cause audible IM distortion, it will instead be clipped at the point determined by the "Protection Threshold" control. Note that if both the "Clip Threshold" and "Protection Threshold" controls are set to the same value, the bass clipper will operate in all cases in a "fixed threshold" traditional manner.

Punch threshold and duration

The "Punch Threshold" control allows the initial edge of the bass waveform to be clipped at a different threshold for extra bass punch; the amount of time this threshold is active is determined by the "Punch Duration" slider.

Settings

System

General settings for SST

Auto Start with Windows

Controls whether or not the SST software will start automatically with Windows.

Bypass processing (On/Off)

Disables all SST processing stages.

Enable Hear & Delta buttons (On/Off)

Under normal circumstances, the "Hear" and "Delta" buttons are always active so you can easily isolate processing stages and compare changes. But this means that they will affect the on-air signal if SST is in the program chain.   To help prevent inadvertent mistakes after you are done tuning SST off line, place this control in the "off" position to disable all "Hear" and "Delta" functions.

License

License keys for Omnia SST are generated using the request code from this screen, and are specific to each hardware installation. Once you have purchased the software, copy the request code, and visit the appropriate form for activation.

3001-00070-000 | OmniaSST + μMPX Encoder & Decoder:
https://minnetonkasoftware.com/telosproductregistration/Registration_Omnia300100070000

3001-00085-100: OmniaSST without μMPX:
https://minnetonkasoftware.com/telosproductregistration/Registration_Omnia300100085100

3001-00071-000: OmniaSST + μMPX Encoder & Decoder (Additional Decoder License):
https://minnetonkasoftware.com/telosproductregistration/Registration_Omnia300100071000

3001-00086-100: Omnia μMPX Encoder License:
https://minnetonkasoftware.com/telosproductregistration/Registration_Omnia300100086100

Active License Info

Displays the current license status

License ID
The unique ID of this installation.

Valid till

Omnia SST licenses can be set to expire after a period of time (demo use will cause audio impairments after 30 days. When that happens, an unlicensed version is not desirable for on-air demo) Fully licensed versions will show a date far in the future making the license duration essentially permanent. This field displays the expiration date.

Omnia SST/Micro MPX

Licenses can be issued for Omnia SST (bundled), MicroMPX (encoder or decoder), or SST as a stand-alone without MicroMPX. These switches display the status of each license.

Latency & CPU

The settings in this section will affect how the CPU is utilized, and can be used to tune tradeoffs between quality and latency. These advanced controls should be left to expert users and administrators. Under most circumstances, these controls should NOT be adjusted from their defaults. Please note not all controls are fully detailed here. Examples of exceptions where these controls may need adjustment would be in a virtual machine environment, running multiple instances of SST on a single machine, or when minimum latency is the ultimate goal..

Latency (128-4096 samples)

Adjusts the overall tradeoff between latency and quality. Higher latency will result in higher quality, and vice versa.

CPU Usage

This panel adjusts overall utilization of the system's CPU.

Quality (CPU Load)

Determines the maximum amount of loading SST will attempt to place on the CPU. Lower settings will use less CPU, but will reduce the quality of the processing to some extent. For every 10% the Quality slider is below 100%, the CPU load is reduced by 5%.

Ignore high frequencies (On/Off)

Reduces CPU load by ignoring frequencies above 19 kHz. This filter should have no effect on FM or streaming operation. The CPU load reduction can be as much as 30%. Using this setting, however, will increase the latency.

Threads

Adjusts the number of CPU threads used by SST for processing. Additional threads can result in higher quality, but will increase CPU usage.

CPU Affinities

Manually allocates various components of SST operation to different physical CPUs or CPU cores if necessary. Under normal circumstances these should NOT need to be adjusted.

Low Latency Monitoring

Omnia SST features a low latency processing path for studio monitoring purposes. This can be used to feed studio monitors, announcer headphones or other areas where latency is a consideration.

Latency & CPU

These adjustments determine the amount of latency and how CPU resources are allocated to the low latency processing path.

LQLL Threads

Adjusts the number of CPU threads used by SST for the low latency monitoring path. Additional threads can result in higher quality, but will increase CPU usage.

Latency (128 or 256 samples)

Adjusts the overall tradeoff between latency and quality. Higher latency will result in higher quality, and vice versa.

HQ CPU Load (On/Off)

Improves audio quality in the low latency monitoring path at the cost of additional CPU usage.

CPU Affinities

Manually allocates the low latency monitoring path to different physical CPUs or CPU cores if necessary. Under normal circumstances these should NOT need to be adjusted.

Soundcards

This panel allows configuration of audio resources used within SST and will open the same audio card settings described in the Quick Start Guide for Omnia SST and earlier in this manual.

FM Options

This panel contains various controls related to MPX generation in Omnia SST including pre-emphasis, FM stereo pilot, SCA signals, composite clipping, and RF bandwidth limiting.

Main
Global settings for MPX output

Enabled (On/Off)

Enables or disables MPX output entirely

Lowpass (On/Off)

FM broadcast is inherently limited in frequency response to approximately 15 kHz. In many cases it makes sense to limit the audio bandwidth below this point when generating a composite MPX signal. This switch will enable or disable a lowpass filter prior to MPX generation.

Lowpass frequency

Controls the frequency of the lowpass filter. The default is 15.5 kHz. This control will be greyed out if the lowpass filter is disabled.

Pre-Emphasis
Due to noise floor considerations, FM broadcast requires that the high-frequency energy be radically boosted before broadcast, and then reduced in the receiver. This is done through a process known as pre-emphasis and de-emphasis. The emphasis curve is measured by a time constant in microseconds. US broadcasters typically use 75 microseconds (which provides a more aggressive high frequency boost) while other areas of the world use 50 microseconds.

Pre-emphasis time (0/50/75 µs)

Selects the time constant of the pre-emphasis curve applied to the audio prior to the final clipper. Setting this to 0 will bypass pre-emphasis. Transmitted audio will sound extremely dull unless pre-emphasis is done elsewhere (such as in the transmitter).

Pre-emphasize output

Controls whether not pre-emphasis is applied to the FM output. This control is greyed out when pilot generation or RDS generation are turned on, since it is generally assumed you would be applying pre-emphasis to a composite MPX signal which contained a stereo pilot or RDS carrier.

Non-phase linear deemphasis

Applies de-emphasis to the signal. This control is greyed out when "Pre-emphasize output" is on, or stereo pilot or RDS generation is enabled since (as with the pre-emphasize output control) it is assumed that the output will have pre-emphasis applied.

Stereo Generator

Global settings for the 19 kHz pilot and 38 kHz stereo subcarrier

Encode Stereo (On/Off)

Enables or disables both the 19 kHz pilot and the 38 kHz stereo subcarrier.

Pilot injection

Controls injection level of the stereo pilot.

BS412

In several European countries, FM stations must comply with the ITU-R BS412 standard. This standard specifies that the total power over any 60 second period is below a certain threshold. The BS412 limiter in Omnia SST attempts to achieve this without compromising audio quality.

Since the BS412 limiter performs its power evaluation based on audio input over time, you will get the best result if the input level is already nearly constant. The BS412 limiter in Omnia SST knows how to handle pre-emphasis and slightly increased bass. Other filters in Omnia SST will respond less to bass due to ITU-1770 when BS412 is enabled, but certain extreme settings may still cause problems.

When BS412 is enabled, try to avoid using bass boost, or excessive boost in the equalizer of the multiband compressor.

General

Global settings for the BS412 limiter

BS412 (On/Off)

Enables or disables the BS412 limiter.

Target maximum level (60s)

Determines the maximum BS412 output level. In most countries where BS412 is used, this must be set to 0 dB, but in some cases it's +3 dB. When changing this setting, you may also need to adjust input drive to the BS412 limiter.

Max MPX deviation

Lowers the maximum MPX deviation when using BS412. If you have a sound card that's not perfectly linear, you can reduce this setting to lower the maximum peak level of the output. Essentially this control can be used to protect against 75 kHz overshoots with the BS412. Stations that want to sound very dense or "smashed" can lower this setting further to increase the amount of clipping.

Headroom

Determines how much headroom the BS412 limiter has for errors. When lowering this value, the average output level will drop to some extent. If this value is too low, however, sudden volume drops can occur.

Input Signal Strength

Controls the input signal level to the BS412 limiter.

Drive

This value should be set such that the signal is just above the BS412 target level.

Behavior

Controls the BS412 long-term limiter. The long-term limiter adjusts the level very slowly based on 20 different algorithms that predict future input. If the amount of limiting is less than about 1 dB, it should normally not be noticeable.

Attack Shape

Determines how fast the level is dropped if it exceeds the target maximum level in any 60-second period. Reduce this value if the audio tends to change frequently in level. For poorly controlled music with rapidly changing levels, a value of 2 might be appropriate. For "normal" music, a value of 3 should work. If your audio has no dynamic range and is smashed to the wall (or if you happen to broadcast pure tones) go ahead and set the value even higher. It won't sound great (unless you happen to like your audio levels smashed) but it will keep the audio within the BS412 parameters.

HF Limiter Threshold

Sets the threshold at which high frequencies (including pre-emphasis) will be reduced if they would cause the BS412 limit to be exceeded.

LF Limiter Threshold

Sets the threshold at which low frequencies will be reduced if they would cause the BS412 limit to be exceeded.

Bass Coupling

Adjusts the "split" between wideband and bass action in the BS412 limiter when the LF limiter threshold is exceeded. Lower values are wideband (which can result in "pumping" with bass) and higher values will result in more reduction of bass.

RF Bandwidth Limiter

ITU-R SM.1268 compliance settings. The ITU-R SM.1268 (Stokkemask) standard is a mask on the RF spectrum. Enabling this mask makes the RF spectrum less wide, which reduces multipath distortion and improves reception. Using Stokkemask is mandatory in the Netherlands, but it can improve reception in other cases as well.

RF Bandwidth Limiter (On/Off)

Enables or disables the ITU-R SM.1268 (Stokkemask) RF bandwidth limiter.

Allow overshoots

Allows exceeding the mask by small amounts. Can improve audio quality.

Clipping

Controls for the composite clipper portion of the stereo generator, including SSB stereo generation. These controls can both increase loudness and improve multipath performance.

Composite Clipping

Global settings for the composite clipper

Composite Clipping (On/Off)

Enables or disables the composite clipper. Enabling the composite clipper will increase CPU load.

Extra composite clipper drive

Further increases the input level to the composite clipper beyond any boost provided by previous stages of processing.

Clip demodulated at

Some radios may have an issue when demodulating peaks beyond a certain point. Adjust this control to limit the maximum demodulated peak levels. Generally settings between 110% and 130% will have little impact on the audio.

Stereo Coding

Enables various styles of Single Sideband (SSB) operation in the 38 kHz L-R stereo subcarrier. This can improve multipath performance in some cases, but some receivers may not respond well. Experiment with this setting to see what works best in your environment.

Normal – Dual sideband
Fully symmetrical operation of the 38 kHz subcarrier. This is the default mode of operation.

SSB - Standard
"Typical" implementation in most audio processors. Very low frequencies (up to about 150 Hz) are modulated normally, anything higher than that is modulated only on the lower sideband.

SSB - Smooth 12 kHz
A special implementation to improve SSB stereo operation. When a tone of 12666.7 Hz is included in the L-R signal, the upper sideband center frequency (50666.7 Hz) is exactly a multiple of the lower sideband center frequency (25333.3). In the RF spectrum, the combined energy of these two frequencies creates much higher peaks.

If the L-R signal contains a tone at 1000 Hz, it will be modulated at 37000 and 39000 Hz. Switching to SSB modulation would create signal only at 37000 Hz, which causes higher peaks in the MPX spectrum and can be more likely to cause reception problems.

Smooth 12 kHz mode reduces the level of MPX frequencies above 50667 Hz, but leaves lower frequencies as they are (dual sideband modulated).

SSB – Steep 12 kHz
Places a steeper filter at 50667 Hz in the MPX baseband, but otherwise similar to SSB - Smooth 12 kHz.

L-R Asymmetry

Unique to Omnia.SST, this mode allows dynamic adjustment of asymmetry in the L-R sub-carrier. In some cases traditional DSB results in higher MPX peak levels, and in other cases SSB will have higher peaks. L-R asymmetry will perform "opportunistic SSB/DSB switching" to minimize the peaks in the MPX waveform. This alone does not gain much loudness (about .1 dB), but it minimizes distortion prior to the clipper, allowing the clipper to be driven slightly harder. The end result is approximately a 1 dB gain in loudness.

Note that when using MicroMPX with this mode, the bitrate should be increased by approximately 64 kbps (for example, 384 rather than 320).

L-R asymmetry (On/Off)

Enables or disables automatic asymmetry adjustment in the L-R subcarrier. This mode is disabled when SSB is enabled.

Maximum asymmetry

Similar to the various SSB modes, adjusts the maximum amount of asymmetry that will be applied to the L-R subcarrier when in this "dynamic" SSB/DSB mode. As with SSB operation, settings that are too high may cause performance issues with some receivers.

Multipath Stereo Enabled (On/Off)

Enables or disables a clipper mode designed to reduce multipath in some cases.

SCA

Omnia SST includes dual aural SCA generators. These require a special FM receiver and are useful for broadcasting closed circuit services such as reading for the blind services, an IFB/talkback for remote broadcasts, or metering telemetry from the transmitter. Frequencies up to 96 kHz are supported with a 192 kHz sound card.

Use the soundcard options to define audio inputs for these SCA signals.

SCA 1/2 Enabled (On/Off)

Enables or disables each of the internal SCA generators. Associated controls will be greyed out when this is turned off.

SCA 1/2 Injection

Controls the injection level of each SCA. Adjust as necessary using a modulation monitor capable of decoding SCA signals, or with no modulation present. Typical injection level would be approximately 10-20%.

SCA 1/2 Frequency

Select the frequency for each SCA as appropriate. Typical values are 67000 Hz and 92000 Hz.

SCA Lowpass

Applies a lowpass filter to the SCA audio input. 7000 Hz would be a typical value. If this is set too high, interference to other subcarriers (such as RDS) and crosstalk into the main channel audio can result.

RDS

Omnia SST includes a fully-featured RDS (RBDS) generator, capable of the same powerful dynamic RDS functions found in many dedicated hardware RDS generators. This panel controls all settings related to RDS.

Power Button

Enable or disable RDS generation.

Main

Global settings related to RDS generator operation

Use secondary input as external RDS input (On/Off)

If enabled, an RDS subcarrier from an external generator connected to the defined secondary audio input will be used rather than the internal RDS generator.

RDS Injection

Sets the level of the internal RDS generator. Typically this would be 4-5%.

Quadrature

Encodes RDS in "quadrature" mode. This lowers the total levels of the stereo pilot and RDS subcarriers which can allow slightly more loudness.

Texts

Defines the text to be sent in the PS field (static or scrolling) and in the RadioText field (if enabled).

RDS PS Text

The Program Service text (8 characters-can be static or scrolling)

RDS RadioText On (On/Off)

Enables or disables sending the 64 character RadioText field.

RDS RadioText

Defines the string to be sent in the RadioText field. This feature is not supported by all radios, but is often a duplicate of title and artist information scrolled in the PS text field.

Formatting the PS and RadioText fields

According to the original RDS specification, the PS text should be static, and its intended use is to display the station name. However, most stations use this text to broadcast lots of things beside the station's name, such as traffic information, phone numbers, even advertisements. Please be aware that certain receivers - mainly in cars - do not display quickly changing texts because it is seen as a danger and could distract the driver.

Omnia SST can broadcast a constant (not changing) text, as is officially mandated by the RDS specification, but it also supports alternating texts, scrolling texts, and dynamically inserted texts such as the name of the current song.

Note that the RDS transmission speed is very low (at most 22 characters per second) so sending RDS information can take some time. Because the RDS signal is injected at a relatively low level, bad reception, especially for moving car radios, often leads to missing portions of text. This means that even when the reception is good, text need to be transmitted at least two times to be displayed. Many radios even ignore text (or portions thereof) that is not transmitted at least two times. This lowers the effective transmission speed to 11 characters per second, so it can take a lot of time before all the text is displayed properly. The data stream is also shared between different types of data (RDS PI code, RDS PS text, and RDS RadioText, for example). The Omnia SST RDS encoder contains an automatic priority mechanism, which lets text that has just been changed take more bandwidth.

The descriptions below are valid for both types of text, although some (such as scrolling) are not useful for RadioText due to the slow transmission speed.

Many radios can only display a limited set of characters. Generally this consists of the letters A-Z (only capitals/uppercase are supported by some radios), the numbers 0-9, and some special characters such as /, \, -, < and >. This is mainly an issue for PS texts, because they are often displayed on cheap LCD displays on car radios. So especially for PS texts, only use the characters listed above. Note that lowercase characters are automatically converted to uppercase by many receivers, though this can also be done in SST.

Constant Text

To transmit a constant text, just type that text in the appropriate field (PS Text or RadioText) in the interface.

The PS Text must be at most 8 characters (but can optionally scroll) and the RadioText can be up to 64 characters. Any extra characters are ignored.

Example:
POWER-FM

To show different alternating texts, write all the texts you want to be displayed in the appropriate field in the interface. Split the different parts with a / character.

Example:
POWER-FM/88.5 FM
Note: The period character will be replaced by a space on radios that cannot display it.

By default, every part of the text will be displayed for 5 seconds.

To change the amount of time a text is displayed, put the requested time in front of the text, as follows:

1.5s:POWER-FM

Displays the text POWER-FM for 1.5 seconds. When specifying a time in seconds, Omnia SST will automatically make sure that the text is transmitted at least twice so radios can receive it properly, even if that takes longer than the specified time.

3t:POWER-FM

Transmits this text 3 times. Omnia SST will transmit this text 3 times, then switch to the next text. Note that it is possible to use 1t here, but it will lead to poor display on many radios.

Example:
3s:POWER-FM/1.5s:88.5 FM
Displays POWER-FM for 3 seconds, then 88.5 FM for 1.5 seconds.

Scrolling Text

Due to the slow transmission times, scrolling is only useful for PS text. Transmitting a PS text takes a short amount of time (2 transmissions * 8 characters / 22 characters per second) = 0.72 seconds. For RadioText, that's 2 * 64 / 22 = 5.8 seconds, which makes scrolling useless for that field.

For left scrolling, add a < character at the front of your text. To scroll at higher speeds, add multiple < characters. To scroll right, use the > character.

Examples:
<THE BEST HITS IN TOWN
<<THE BEST HITS IN TOWN AT HIGH SPEED
3s:POWER-FM/1.5s:88.5 FM/<THE BEST HITS IN TOWN

The scrolling speed can also be adjusted using timing settings:
3t:<<THE BEST HITS IN TOWN

In the previous example, the first and last characters are displayed only very briefly. There are two possibilities to solve this.

Adding spaces:
3t:<<   THE BEST HITS IN TOWN   (ends here)

It is also possible to use different timings for different portions of the scrolling. That makes it possible to display the start and end for a longer period of time than intermediate parts.

Instead of using just 2s: which sets the display time for every part of the scrolling, different values can be set for different positions.

During scrolling, the first position is called '1', the second '2' etc. The last position is also known as '-1', the one before that as '-2', etc.

One scenario might be to specify that positions '1' (start) and '-1' (end) display for 1.5 seconds, and in addition that everything in between (2 up to -2) should be broadcast 2 times (the minimum for good reception). That can be done as follows: 1=1.5s,2..-2=2t,-1=1.5s:

Examples:
1=1.5s,2..-2=2t,-1=1.5s:<THE BEST HITS IN TOWN
3s:POWER-FM/1.5s:88.5 FM/1=1.5s,2..-2=2t,-1=1.5s:<<THE BEST HITS IN TOWN

Word wrapping text

Instead of using scrolling, you can also use word wrapping. This way you can type a longer text, and the words will all be displayed separately.

Word-wrapping is enabled by adding two pipe || characters.

Example:
||THIS TEXT IS WORD WRAPPED

Again, it's possible to adjust the time that a word is displayed:
1.5s:||THIS TEXT IS WORD WRAPPED

If a word does not fit on the display (if it's longer than 8 characters), the remaining characters are truncated. To avoid that, you can combine word wrapping with scrolling.

Combining word-wrapping and scrolling

If you combine word wrapping with scrolling, every word in a text is displayed separately. If it does not fit on the display, scrolling is used to display the whole word.

Example:
 ||<TRANSMISSION IS A LONG WORD

Now the scrolling problem from before re-appears: The start and end of the word are displayed only very briefly. This can be solved using the timing settings described for scrolling.

So, to display the text, every word 1.5 seconds, but if a word scrolls its start (1) and end (-1) should be displayed 1 second, and the parts in between (2 up to -2) should be broadcast 2 times, that leads to: 0=1.5s,1=1s,2..-2=2t,-1=1s:

Example:
1.5s,1=1s,2..-2=2t,-1=1s:||<TRANSMISSION IS A LONG WORD

Including dynamic texts

Dynamic texts can be used to include all kinds of changing information in your RDS texts. For example the name of the song that's currently playing, the current temperature, the weather forecast, you name it.

Omnia SST enables you to include this type of data, as long as you can make the text available in a file. If you have that, there are 4 ways of including data from a file into your RDS text:

\R"C:\Filename.txt"
Reads raw (unformatted) text from C:\Filename.txt, converts it to capitals, and inserts it here in the RDS text.

\r"C:\Filename.txt"
Similar to \R, but does not convert the text to capitals. If your input file contains lowercase characters, some radios may not display parts of the text.

\F"C:\Filename.txt"
Reads formatted text from C:\Filename.txt, converts it to capitals, and parses it as if it were a part of the filled-in RDS text. Any special characters (<, ||, : etc.) are used for formatting.

Only use this if you know what you are doing. Unexpected special characters in the file can lead to text display oddities.

\f"C:\Filename.txt"
Similar to \F, but does not convert the text to capitals. If your input file contains lowercase characters, some radios may not display parts of the text.

Unless you have a specific reason to do otherwise, you should only need \R.

Example:
POWER-FM/PLAYING/\R"C:\NowPlaying.txt"

The text files are read from disk each time when the sequence repeats. Using a local disk is recommended so everything gets cached. If you use a remote disk, the process might stall until it can access it. Do not use a network disk!

Because of this possibility, there are 10 seconds between the moment when the file is read and the moment when the contents of the text file start to be transmitted. To avoid that, place the dynamic part at the beginning of the sequence - it repeats anyway:

\R"C:\NowPlaying.txt"/POWER-FM/PLAYING

Because song names can be long, you'll probably want to use the word wrapping and scrolling trick described above:
1.5s,1=1s,2..-2=2t,-1=1s:||<NOW PLAYING \R"C:\NowPlaying.txt"ON/6s:POWER-FM

How can I get the name of the current song in a file?

Most major automation systems can provide an output to a text file for now playing data. Check with the manufacturer of your automation system for further information on how to do this.

Special characters

As you can see above, some characters have special meanings in the Omnia SST RDS generator. More specifically, the characters <, >, |, :, / and \ cannot be used in the RDS messages without special considerations.

If you do need to use any of these characters, put a \ in front of it. So: \<, \>, \|, \:, \/ and \\.

Example:
\<POWER!\>

Info
Various RDS metadata flags describe characteristics of the programming and RDS data stream. Some of these are not likely to be implemented on many receivers.

RDS PI

Enter the PI code of your radio station.
The PI code controls things like automatic switching to Alternative Frequencies. It's important to set this up correctly. Here's why:

If you use the same code as another station in your area, you might inadvertently 'steal' listeners from them (the radio might automatically switch to your station if its reception is better) or you might lose listeners to them in areas where their signal is better. This is extremely annoying to listeners.

If you change your code at a later time, some radios where your station has been programmed on a preset button may turn the sound off because they 'think' it's the wrong station, even if reception is perfect. Only reprogramming the preset button helps in that case - but not all listeners will realize this.

The PI code needs to be entered as a hexadecimal number. The default is FFFF, which means that it should be disabled - but not all receivers adhere to this.

The first 2 digits of the PI code are generally a country code, but again, many receivers ignore it. Whatever you do, make sure that the last 2 digits are unique in your area, and that they are not FF.

That said, PI codes are regulated differently in different countries. Please check with your appropriate broadcast authorities what you need to do. In the US, the PI code should generally correspond to your call letters.

A handy calculator can be found here: http://www.w9wi.com/articles/rdsreverse.htm

PTY

The type (format) of program that you're broadcasting.

Many receivers can automatically search for some type of broadcast, such as a news broadcast. If you don't always broadcast the same type of program and you want to update this setting, make sure that Dynamic PTY is enabled.

The list of PTY codes is very different for RDS (EBU, used in Europe) and RDBS which is used in the US. The PTY codes are changed when you change the Pre-emphasis time.

Dynamic PTY (On/Off)

Enable if the PTY changes regularly.

RDS TP (On/Off)

Indicates that the station regularly broadcasts traffic bulletins.

RDS TA (On/Off)

Indicates that the station is CURRENTLY broadcasting a traffic bulletin.

RDS Music (On/Off)

The station broadcasts music (Not frequently used)

RDS Artificial Head (On/Off)

Indicates a binaural head was used for the recording (Not frequently used)

RDS Compressed (On/Off)

Source is compressed (Not frequently used)

RDS ClockTime (On/Off)

Broadcasts the current date and time once per minute in the RDS data stream. Receivers supporting this feature will automatically set their clocks by this data. If used, make SURE that the PC running Omnia SST has an accurate source of time data.

Alternative Frequencies

Define up to 25 alternate frequencies that this station's programming is carried on. Generally used in European broadcast networks, and not consistently implemented in the US. Some radios use signal strength when scanning for AF, others appear to use the PI code. Only utilize these fields if you understand that receiver behavior with the "AF" feature will vary depending on a number of factors. Some receivers ignore AF entirely.

RDS UECP/Remote control over TCP/IP

UECP is a standardized protocol that is used to send information to RDS encoders. It is supported by many software and hardware RDS encoders, and by many automation systems.

Originally, UECP was used over an RS232 connection, but many products are now supporting it via IP. Omnia SST only supports the latter.

Enable UECP (On/Off)

Enables or disables control via UECP.

UECP is connected (On/Off)

(read only) Displays the current status of the UECP connection (On = connected, off = not connected)

Mode (UECP/ASCII)

Selects between control via UECP or ASCII. Many applications now support UECP directly, but for those that do not, an ASCII mode is supported for sending commands in plain text.

Port

Specifies the IP port where Omnia SST should listen for RDS encoder traffic.

Specific Address

UECP can send commands via multicasting to multiple RDS encoders. If this feature is utilized, this addressing information can be used to target a subset of encoders.

Specific Address (On/Off)

Listen only for UECP commands targeted to the specified address.

Site/Encoder Address, DSN, PSN

Enter the information appropriate to your RDS network if this feature is utilized.

Error behavior

Controls what the encoder does should it lose a UECP connection

Use standard RDS settings when connection closed (On/Off)

Fallback to the RDS settings specified in the "RDS" section of this panel should the UECP connection drop. It is a good idea to enable this and populate the RDS settings with "generic" information should the automation system fail to connect and provide data via UECP.

µMPX

MicroMPX or "µMPX" is an Omnia exclusive, and allows transmission of an entire composite MPX baseband via IP in streams as low as 320 kbps. Omnia SST is the first product to feature this technology. The Omnia SST GUI includes controls for a MicroMPX server. A standalone decoder application is used as the client, and a stand-alone Encoder is also available . (see appendix later in this document for more information)

Power button

Enables or disables MicroMPX encoding and transmission from Omnia SST. µMPX

Global settings for the MicroMPX server

Bitrate

The recommended bitrate (and the default) for MicroMPX is 320 kbps.   Lower bitrates will decrease quality, while higher bitrates will increase quality at the cost of bandwidth use. Bitrates from 300-400 kbps are available, 320 kbps and higher are supported.

Reset packet interval

The MicroMPX server will periodically send "reset packets" to synchronize the client. Depending on the stability of the connection, it may be necessary to send these packets more frequently. On most connections, the default (and maximum) of 5 seconds is generally sufficient. Decrease this value to send reset packets more frequently.

Streams

This section configures the destinations for the MicroMPX streams. Multiple destinations are supported simultaneously.

On/Off

Enables or disables sending MicroMPX traffic to each destination individually.

IP

Specify the IP address of the host running the MicroMPX client. Multicast and broadcast IP addresses are also supported (any valid multicast IP, or x.x.x.255 for broadcast to an entire subnet).

Port

Specify the IP port on which MicroMPX traffic will be sent. The default is port 8854 and generally should not need to be changed unless some other application is utilizing port 8854.

Error Correction

The MicroMPX server incorporates an advanced error correction scheme which can send error correction packets along with the audio, with a tradeoff of increased bandwidth use. These settings adjust how frequently these packets are sent, and how many are sent in each group. A bandwidth limiter is also available to prevent error correction overhead from exceeding available bandwidth.

Error Correction Span (delay)

Adjusts how frequently the error correction packets will be sent. A setting of 0 will disable sending of error correction packets. Smaller values will send packets more frequently and consume additional bandwidth.

Error Correction Overhead

Adjusts the number of error correction packets sent at each interval. Larger values will consume more bandwidth. As with the error correction span, a value of 0 will disable sending of error correction packets.

Rate Limiter (On/Off)

Enables or disables rate limiting for the error correction packets. This setting can assist with bandwidth management when utilizing error correction packets. While it will not prevent you from exceeding available bandwidth (SST really has no way of knowing what your connection can provide) it will allow you to "fine tune" the amount of bandwidth used for error correction.

Limit Rate Below

Attempts to limit the bandwidth use (including error correction) below the specified value. This value will change dynamically depending on the error correction span and overhead settings, so use this setting to predict approximately how much bandwidth your MicroMPX stream will ultimately consume when applying error correction.

Presets

Like most other audio processors, Omnia SST includes a system for managing presets. These presets cover several different categories of parameters including global I/O settings, the "Cleanup" sections, and the processing sections.

Click the "Presets" button on the navigation bar on the access the preset library.

Depending on which portion of the interface you are in, the available list of presets will change. For example, if you click on "Presets" from the "FM Options" page, you'll see IO configuration presets. To see the entire list of presets for all categories, start from the "Home" screen.

Loading a preset

The preset window is split into several categories: Recent, Processing, and Clean up. "Recent" will display the most recently loaded presets from any section. The "Processing" and "Clean up" categories contain presets specific to those areas.

If you attempt to load a preset from an area of the software where there are no presets defined, you will see a message indicating "You have not saved any presets that contain configuration settings yet".

To load a preset, simply click it. The sections available to load from that preset will be displayed. Click the appropriate button for the section to apply that preset to (Cleanup, Processing, Full FM preset, All settings).

Note: The currently loaded preset will be highlighted by a yellow frame. If sections from multiple presets are loaded, multiple presets may be highlighted.

Save Current

Click "Save Current" to save a custom preset based on the current settings. This will save parameters from all sections simultaneously.

Enter the desired name for the preset, along with any other descriptive data (preset author, intended preset format, description) then choose to save as a new preset or overwrite the currently loaded preset.

Note: To keep presets from being deleted by mistake, there is currently no way to DELETE a preset from within the interface. Preset files are stored in the system's "Users" directory for the currently logged in user under "OmniaSST.Presets". The list of available files is read when SST starts. If you wish to delete presets that you no longer use, such as iterations spawned in the process of creating a master preset, use windows explorer to find and delete old presets, then reload the SST application

Import

Click the arrow to import a preset from a file.

Search

Click the magnifying glass to search the list of available presets.

Close

Click the "X" to close the preset window.

Metering

Metering is available in two forms: A dedicated meter panel displaying all available meters, and a meter banner which displays metering specific to whichever processing section is being adjusted. The following is a list of what meters are available, and their function is generally self explanatory.

Input Scope

Displays the input waveform. Audio detected as clipped will be displayed in red.

Declipped Scope

Displays the output waveform of the declipper.

Output Scope

Displays the final output waveform after all processing stages.

MPX (Baseband spectrum analyzer)

Displays the MPX baseband output including the stereo pilot, RDS and any other subcarriers.

RF (RF spectrum analyzer)

Displays an approximation of the frequency modulated RF envelope (as would be seen on a spectrum analyzer set to the carrier frequency of the station).

Left Demodulated Output

If enabled, displays the demodulated output from the FM section of the processor. In essence, this would be similar to looking at the output from an FM receiver.

Input Meter

Displays the input audio level. Actual input level will be displayed in light green, while gain added in soundcard configuration will be displayed in darker green.

Output

Displays the output audio level. The light part is after any added gain. The dark part reflects the level after processing and must not exceed the top (0 dB).

MPX Power

Reflects the MPX power level of the FM signal in dBr.

FM

Displays MPX output level, as well as peak demodulated left and right levels.

Delossifier

Indicates the likelihood that the input audio is from bit-reduced source material (MPEG L2, L3).

Noise Gate

Shows noise reduction (gating) per band.

Natural Dynamics

Indicates the activity of the "Natural Dynamics" section. Light grey indicates the amount of expansion, dark grey indicates the total amount of possible dynamic expansion in each band.

Phase

Displays the amount of phase correction applied by the "Azimuth" section.

Auto EQ

Gives an overview of the amount of boost or cut being applied to each band by Auto EQ. Grey indicates boost, red indicates cut, and purple lines indicate the average boost or cut recently applied in each band over time.

AGC

Shows the amount of gain reduction occurring in the AGC (Automatic Gain Control) section. For most "Dynamics" sections (AGC, Multiband, Singleband), the action of both "Compression" (Grey) and "Limiting" (Yellow) are displayed, along with "Gating" displayed as a horizontal bar along the bottom in purple.

Stereo

The left meter indicates the amount of "multiband" stereo enhancement, while the right meter indicates the amount of "wideband delayed" stereo enhancement applied.

Equalizer

This display mirrors the one in the Equalizer panel--displaying the EQ curve being applied, as well as an RTA (real time analyzer).

BIYF

Displays the amount of enhancement applied in each "Bass In Your Face" section.

Harmonics

Amount of extra bass harmonics generated in the "Phat Bass" section.

Multiband 1 & 2

Displays the gain reduction of each band in the the two multiband dynamics sections.

SB

Gain reduction applied in the final single band dynamics section prior to the clipper.

Clipper

Indicates RMS gain reduction in the final clipper.

Micro-MPX (μMPX) addendum

Omnia μMPX®

Radio's First Composite FM Codec: High-quality MPX over a 320kbps IP connection!

Thank you for your interest in Omnia μMPX (Micro MPX).

μMPX is a specially designed audio codec that is able to transport high-quality Multiplexed FM signals (including RDS data) over a relatively small 320kbps data pipe. This specialized codec is purpose-built for FM radio. By reducing data requirements, high-quality multiplexed audio can be economically routed from an audio processor, over IP, and directly into an exciter. The potential for this technology is enormous.

Lower bandwidth IP connections and narrow band STL channels can now be put into play to transport MPX signals. Omnia μMPX will create new audio distribution options and cut equipment needs and costs, as it leverages advances in the world's ever-growing IP infrastructure. The inclusion of uMPX into other Omnia processors going forward will open up new, previously unheard of integrated STL transport options!

Omnia SST is the first processor to include integrated μMPX capabilities. An encoder portion is built right in to SST, and a separate windows application acts as a Decoder (receiver). Additionally, a stand alone Encoder is available as a Windows application

This document will help Omnia SST users get a basic send-receive system running in a few steps. For simplicity, we recommend you first get μMPX running on a single PC. This will provide a "proof" case which avoids more complex networking issues. Once you have a "local" implementation running on a single PC, you can apply your broader networking skills to extend operation over a LAN or WAN.

While Omnia μMPX is a separate application capable of being used as a stand-alone codec, for ease of illustration, we will use Omnia SST as the generator of an encoded μMPX signal, and the μMPX standalone decoder app.

Step 1: Get μMPX to work on one pc with standard settings

This step requires a 192 kHz capable sound card

Encoder:

♦ Start Omnia SST.

♦ Click on the Omnia SST tray icon to open the web interface:

♦ Under Settings -> Sound cards, set the sample rate to 192 kHz.
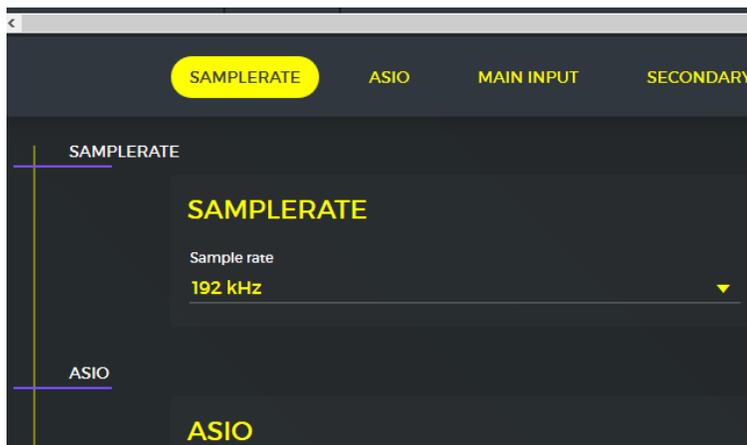
♦ Under Settings -> FM Options, enable MAIN (which enables FM processing), set the Pre-emphasis time as appropriate for your location and at turn on the stereo generator (Encode Stereo). You may also want to enable RDS too, under Settings -> RDS). Make sure that you see the pilot in the MPX display on top of the screen. Audio is also present in the screenshot below:

♦ Go the the Settings -> μMPX page.

♦ Type 127.0.0.1 for the first MPX output, and enable μMPX.   Leave all the other settings at their defaults (320 kbit/s, 5.0 sec Keyframe interval, no error correction, no rate limiting). Note: **127.0.0.1** is the loopback Internet protocol (IP) address also referred to as the "localhost." This address is used to establish an IP connection to another program running on the same computer.
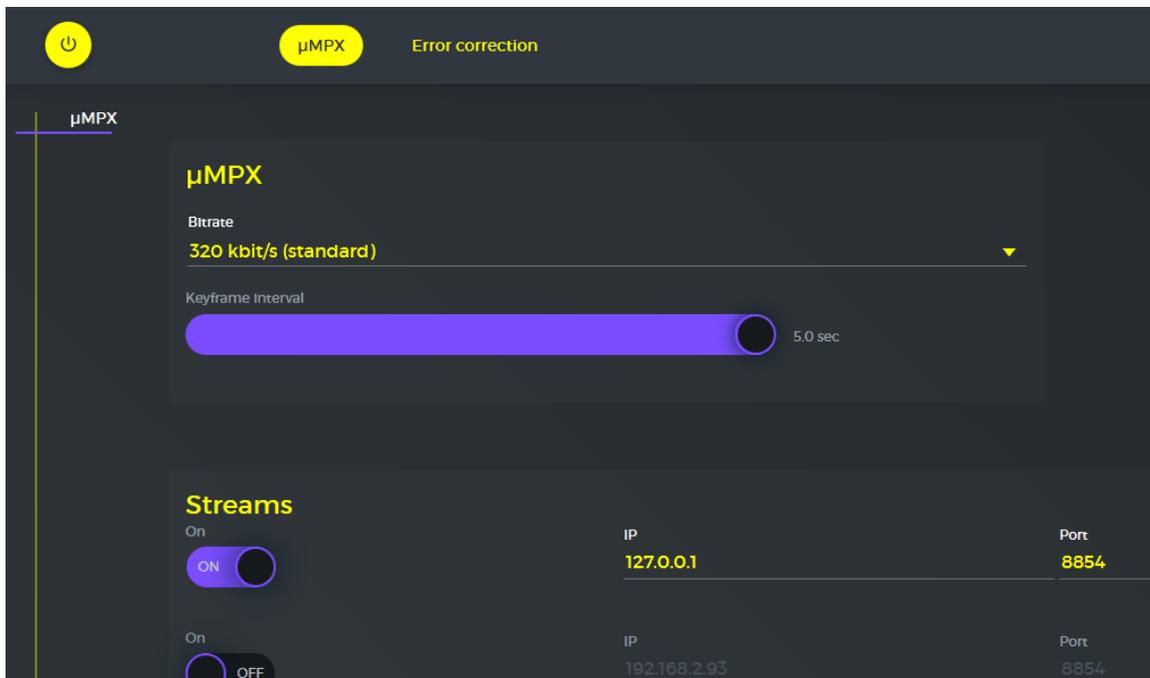
If it's not already on, Turn on the "power" button above the uMPX settings. Congratulations! You're now streaming a μMPX signal and are half the way there! Now that you are encoding a stream, you need to set up a decoder to receive it.

Decoder

The μMPX Decoder is a separate application, designed to provide simple IP and soundcard configuration for receiving a signal from a μMPX encoder, and is engineered to use very little system overhead. SST comes with a single license of both μMPX Encoder and Decoder. Additional, stand-alone licenses are available.

Starting MPX

After configuring your encoder, launch the MicroMpx Decoder executable from the desktop icon, then go to your system tray and click on the small, blue "u" icon to open the browser interface:
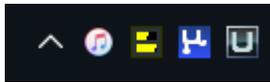
Note that you may need to change the port number from the default (8080)if you have SST running on the same machine or the SST interface will open in the browser instead of the uMPX decoder. Try Port 8081 for the decoder interface.

The μMPX interface will open within your chosen default browser.

There are three choices from the main menu: License, Sound Cards and μMPX:
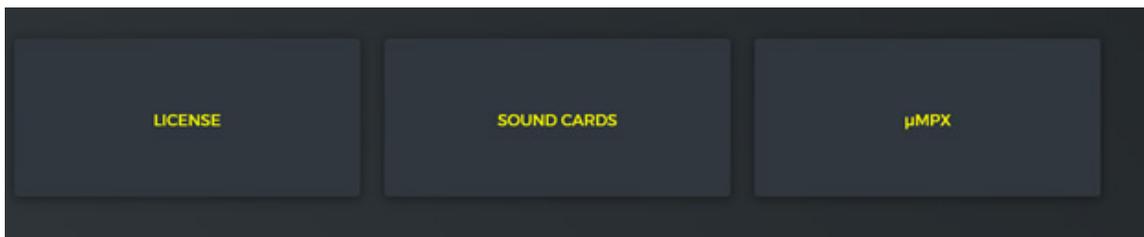
Within the SoundCards and μMPX screens you will see three windows at the top of the screen that show:
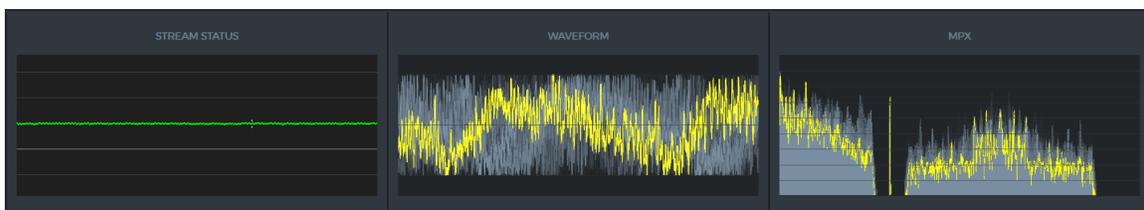
Stream Status Window

When µMPX is decoding a signal, you will notice a green horizontal line indicating the decoder is seeing a µMPX encoded signal from the other source on your network.

Waveform Window

When µMPX is decoding a signal, you will see the waveform of this signal represented in this middle window.

MPX Screen

When µMPX is decoding a signal, you will see the entire baseband signal of the output represented in this window.

Home Button

The home button in the upper left of the screen is a convenient way to navigate back to the three main choices in the main menu.

Warning sign

The triangular warning sign is where µMPX will provide diagnostic information regarding network issues/outages, or issues regarding your sound card configuration or licensing information. If the warning sign glows red, click on the icon to see a listing of issues or errors that µMPX is encountering.

Help is available by pressing the question mark icon in the upper right of your browser, then clicking on an item you need more information on.

LICENSE

License keys for Omnia SST are generated using the request code from this screen, and are specific to each hardware installation. Once you have purchased the software, copy the request code, and visit the appropriate form for activation.

3001-00070-000 | OmniaSST + µMPX Encoder & Decoder:
https://minnetonkasoftware.com/telosproductregistration/Registration_Omnia300100070000

3001-00085-100: OmniaSST without µMPX:

https://minnetonkasoftware.com/telosproductregistration/Registration_Omnia300100085100

3001-00071-000: OmniaSST + µMPX Encoder & Decoder (Additional Decoder License):

https://minnetonkasoftware.com/telosproductregistration/Registration_Omnia300100071000

3001-00086-100: Omnia µMPX Encoder License:

https://minnetonkasoftware.com/telosproductregistration/Registration_Omnia300100086100

Activate License

Once given your code, press Activate License. Your license ID and expiration date will appear below.

FM Output

Choose the 192kHz capable sound card that will be used to feed your FM transmitter or exciter.

Sound Cards
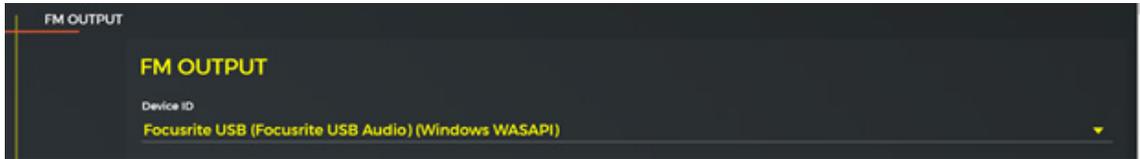
Volume (MPX Level):
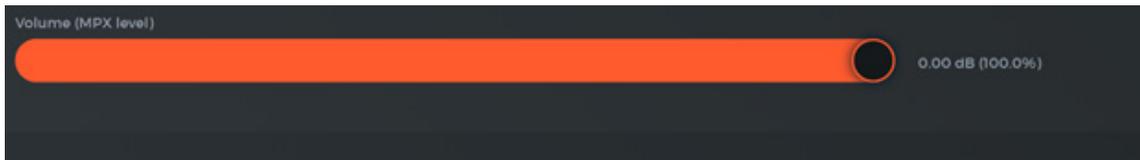
Use this slider at to adjust the MPX output of your sound card that feeds the exciter or transmitter.

FM tilt correction and Test Signal
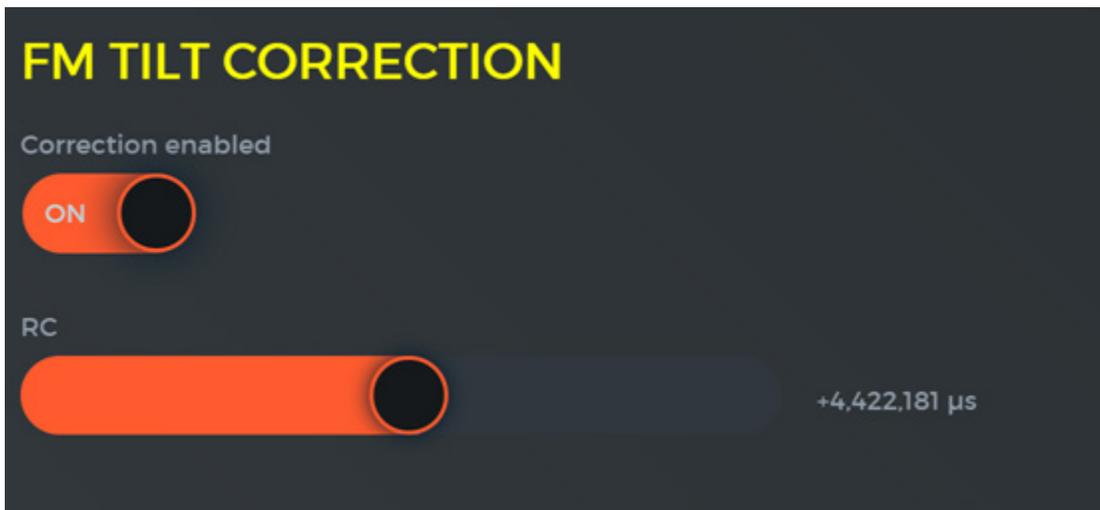
Use these together to compensate for sound cards that are not DC-coupled or might have other non-linearity. To ensure the best results, even sound cards advertised as DC-coupled should be checked for non-linearity before being fed into a transmitter.

To use FM tilt correction with test signals: Turn on both FM tilt correction and test generator. On an external oscilloscope monitoring the sound card's MPX output, or using the onboard , monitor the output of μMPX while choosing Smooth Square from the dropdown menu of Test Signals available on the right side of the screen.

Test Signals
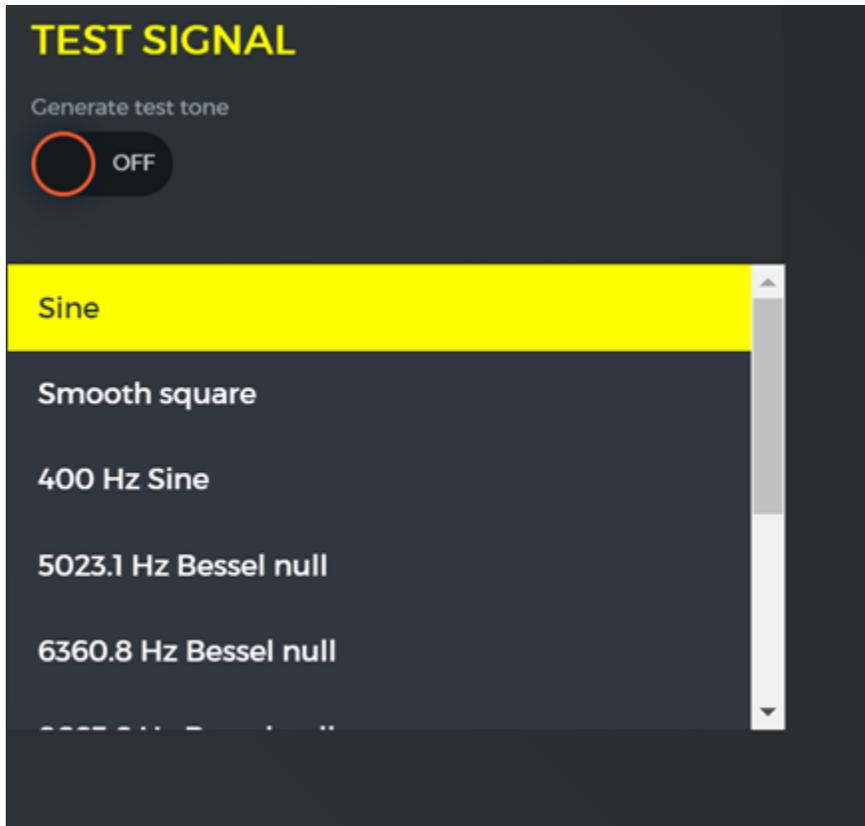
Turning the test generator on will interrupt any feed from the decoder[GS2] and place test tones on the output of the chosen sound card. This drop-down menu provides a number of useful test signals to test the linearity of your sound card and the rest of your transmission chain.

Test signals include: Smooth Square Wave, 400Hz Sine Wave, 5023.1 Bessel Null Tone, 6360.8 Bessel Null Tone, 8665.6 Bessel Null Tone, 13587.0 Bessel Null Tone, and 31189.4 Bessel Null Tone. Sawtooth with Pause and Sawtooth flat bottom are designed to be used for instances of μMPX that may be feeding an AM modulator.

RC Slider
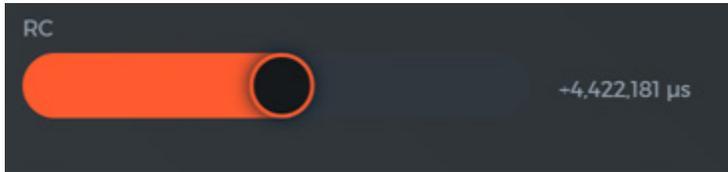
Using the "Smooth Square" signal, adjust the RC slider as necessary to achieve square flat top waveform at frequencies of 30Hz, 400Hz and 10kHz on the test signal generator to the right

MPX Menu
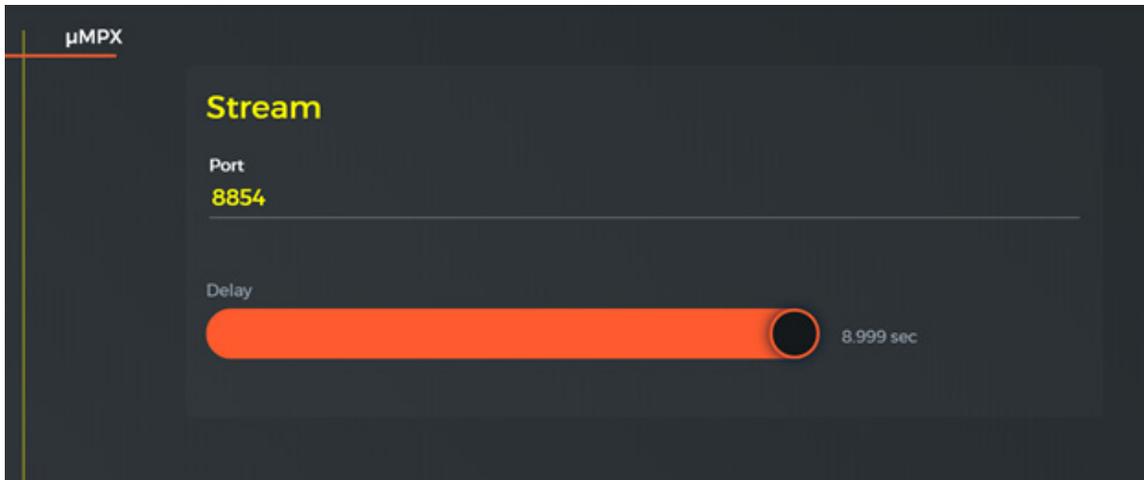
To decode a signal, choose your incoming port number, (default port is 8854) and toggle the "power" button in the upper left hand side of this section.

Delay Slider

Use this control to adjust up to 9 seconds of buffering delay. This may be useful in situations where extra buffering is needed, or if you are trying to time align signals.
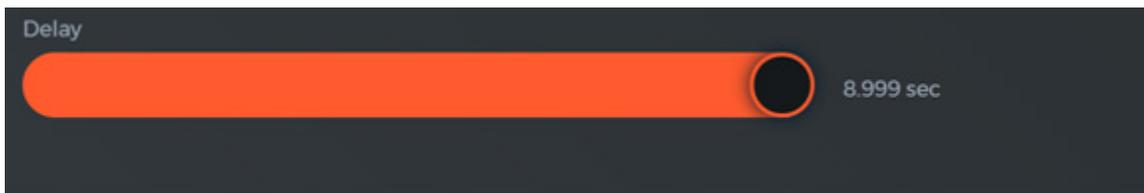
Backup Player
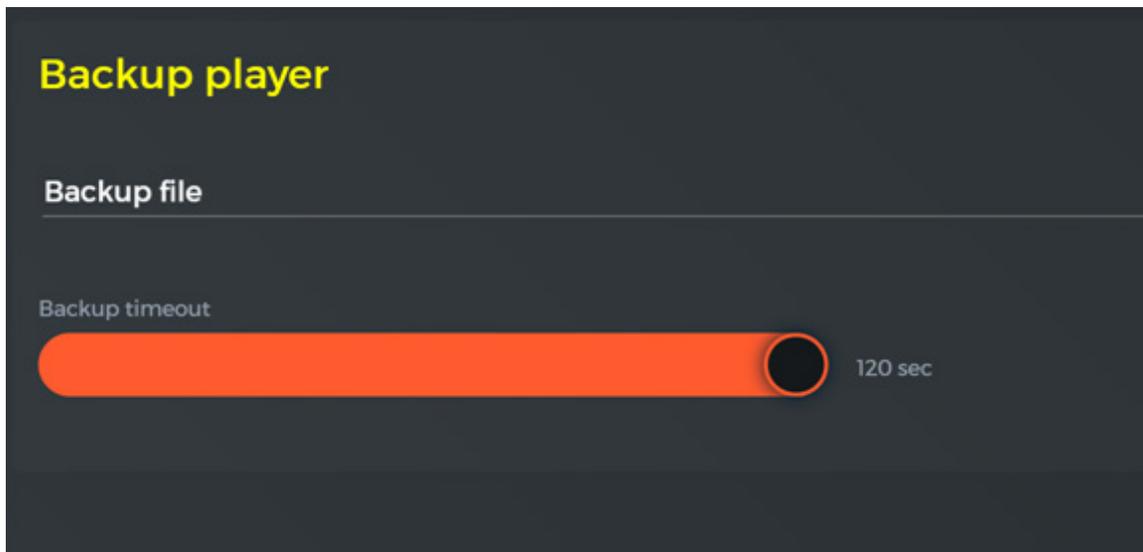
µMPX Decoders include a simple backup playout system. This system allows a single MPX audio file to be played in the event that the encoded stream feeding the decoder is lost. Source audio files must be composite audio (MPX) format. µMPX checks if the file is valid, i.e. contains only one channel at 192kHz. It gives an error if you try to put in a file that does not comply with this. The backup audio file should be stored on the same PC that is running the decoder application. The path location and name (eg; c:\audio\backupfile.wav) of that file should be placed in this window (above).   The file is played in a loop. When the stream comes back and results in properly decoded audio (so at least one keyframe was received), the backup file will fade out and the stream will fade in. If the stream drops out again, the backup file playback will resume playing from where it was last stopped.

Backup timeout
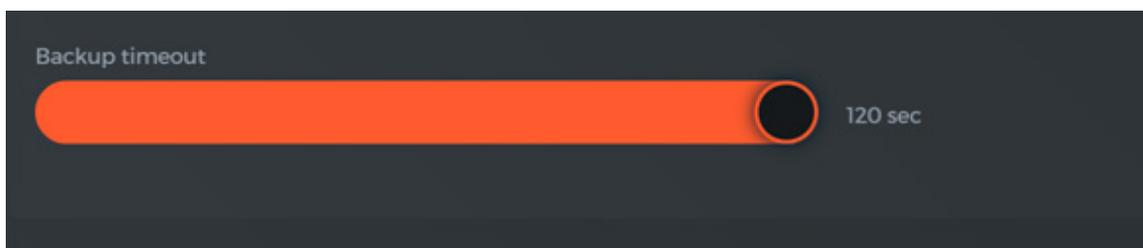
 This control specifies the silence threshold level and timeout before switching to the backup file.

Choose a value by moving the slider between 1-120 seconds of silence before the Backup File begins playing straight to the output of the µMPX decoder's chosen sound card.

Creating a backup file

The simplest way to create a backup file is to record the analog composite output or your uMPX decoder into a sound card that supports analog input. Create a 192kHz mono file, choose audio that is "generic" such as a music bed with the call sign of your radio station. A length of 5 to 15 minutes should be sufficient. While the file wrapper may be saved as a .WAV or .FLAC, note that µMPX will check the file is compatible, since a L/R wav will not be suitable. Be sure to use an editor package that supports 192kHZ.

- ♦ **Standalone GUI versions of the MicroMPX encoder and decoder are installed in C:\ Program Files (x86)\OmniaSST\MicroMPX_Encoder and C:\Program Files (x86)\ OmniaSST\MicroMPX_Decoder.**
- ♦ **ARM versions of these executables are also available under special OEM licensing arrangements. For more information, please contact Telos Support.**

Step 2: Streaming to another pc

Now that you are successfully streaming a µMPX internally in a single PC, you can extend the recipe to stream to a different, external PC, reaching it via its IP address. This external PC will also need a 192K capable sound card, and you will want to have SST installed, but only so you can run the Decoder.

Finding your IP address
- ♦ If you're on a local network or a VPN, you can easily find out the local (network) IP address of the machine you are on by typing "ipconfig" from a command prompt:
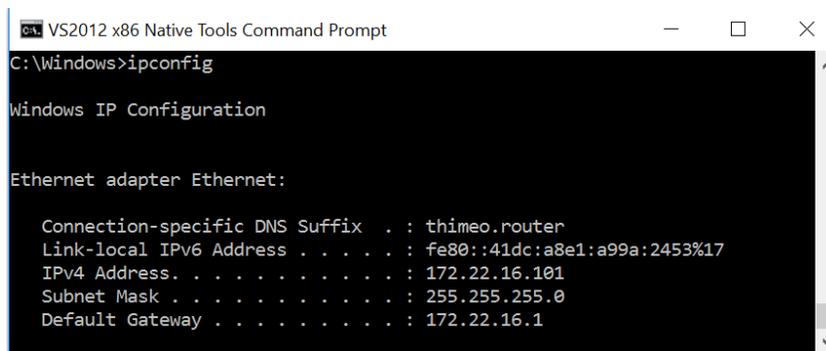
```
VS2012 x86 Native Tools Command Prompt                    —    □    ×

C:\Windows>ipconfig

Windows IP Configuration


Ethernet adapter Ethernet:

   Connection-specific DNS Suffix  . : thimeo.router
   Link-local IPv6 Address . . . . . : fe80::41dc:a8e1:a99a:2453%17
   IPv4 Address. . . . . . . . . . . : 172.22.16.101
   Subnet Mask . . . . . . . . . . . : 255.255.255.0
   Default Gateway . . . . . . . . . : 172.22.16.1
```

In this case the IP address of the decoder is 172.22.16.101   Streaming over a LAN (without going through firewalls or port forwarding) is the next logical test to demonstrate you are sending and receiving the µMPX signal.

(You will also need this info if you plan to use the public internet!)

♦ Returning to your PC running Omnia SST, enter the IP address of the PC you want to stream μMPX to, as in the example below:
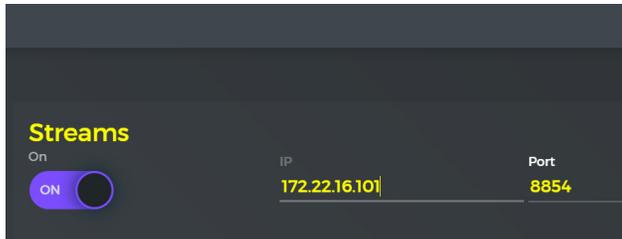
♦ Now, from the second (external) PC, run the Decoder application just as you did for the internal test. Assuming you are running on a LAN, you can setup the decoder in the same way as above, except you are now sending the μMPX signal over a network. Assuming your sound card is properly set on the second machine, you should now be getting an MPX signal out from your audio card, and you're done!

♦ With enough bandwidth, you can send μMPX signals wherever you have a clean IP connection. Whether you are sending over a LAN, a WAN, using a VPN, or even if you are using the public internet, a single Omnia SST system encoding μMPX can be delivered to multiple μMPX decoders on different machines at different destinations.   Your distribution can be as simple as a single point to point connection, or a "one to many" setup with dozens of decoder endpoints. μMPX harnesses standard internet protocols, so possibilities are gated only by what your network infrastructure and IT skills will support. Note however that each Decoder will require a separate license.

PLEASE NOTE!

While it may be possible to use the public Internet with μMPX, we can make no guarantees that performance, bandwidth or up-time will be adequate for your needs.

Low budget outfits that have few or no STL choices may consider the Internet to be perfectly adequate. Major operators with real revenue on the line may consider the Internet a last resort backup link only. Results will vary. The risk is on you.

♦ If your Decoder machine is on the internet, you'll need a public IP address, and you'll probably need to do some port forwarding. The easiest way to find your public IP address is to use a service such as whatismyip.com, whatismyip.org, whatismyipaddress.com, …:
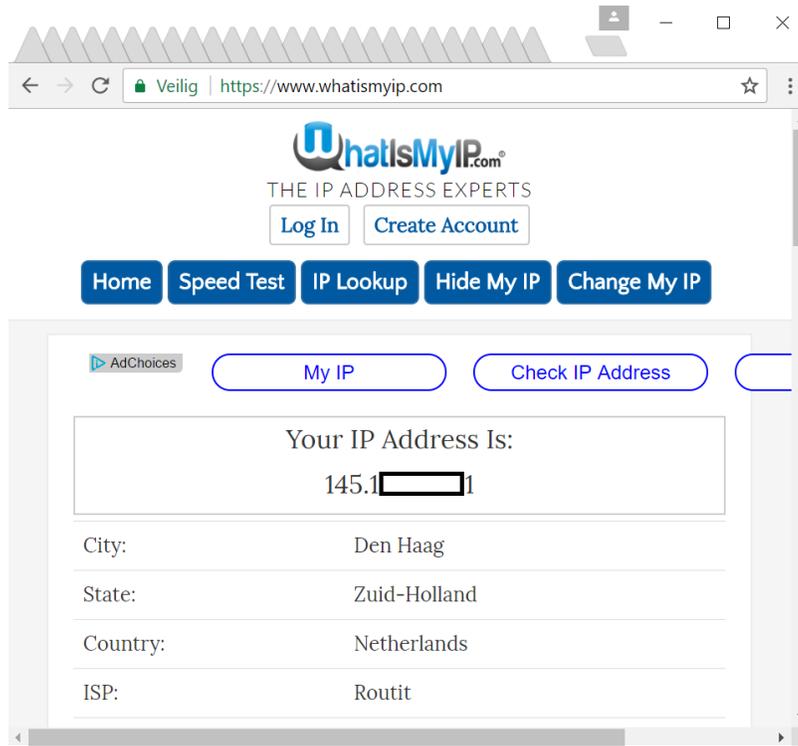
In this case the IP address of the decoder is (example) 145.1nn.nnn.nn1.

♦ If you use a public internet address, and there's a router between your decoder pc and the internet (there probably is), you'll have to enable port forwarding to make the decoder pc reachable by the encoder. Specifics for your network will depend on your router topology. As an example:
Go to your router settings. This is usually at the IP address that's listed as default gateway in the ipconfig output. Typical addresses are:
  192.168.100.1
  172.22.16.1
  10.10.10.1
Log in (log in info may be printed on a sticker on your router, you can try if admin / admin, or consult your network administrator). Then search for menus that allow you to setup "Port forwarding", "Virtual Server" or similar.

Once you find the correct menu, set things up such that the port number that you are using on the sender (by default, port 8854) is forwarded to the local IP address (see ipconfig above) of your decoder pc.
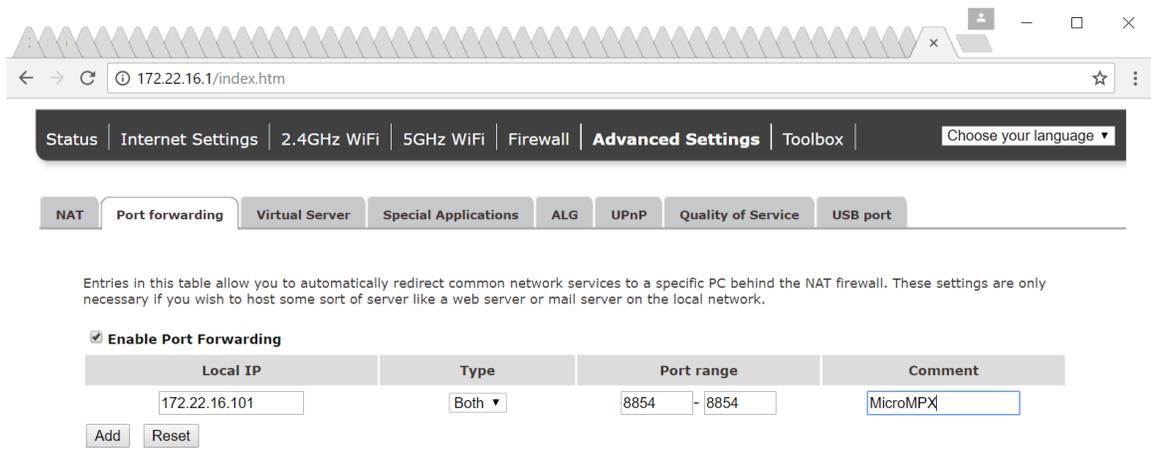
In some routers, if you want to map a port to a different external port number than what you are using internally (for example, if you need a port that is already in use, or if you want to send multiple µMPX streams to the same network), you might need to use a 'Virtual Server" instead of "Port forwarding". In the example above, you may not be able to enter what a certain port should map to, so if you want map ports 8854, 8855, 8856 and 8857 to different decoders on your local network, you may need to use Virtual Server instead.

For comprehensive notes of how to setup port forwarding on nearly any router imaginable, go to http://portforward.com/router.htm.

Step 3: Gain and tilt

With option -g or --gain, you can change the level going into the transmitter. Set this such that your modulation level is correct (typically 100%).

MicroMPX -c 8854 -a "Speakers" -g .521

With --tilt, you can correct for a sound card (or transmitter) highpass filter that would cause a tilt in the output signal. The value to supply here is identical to the value in OmniaSST. Options available in SST to find out how much tilt correction is needed, so please use OmniaSST to determine that value (if needed). With the recommended sound cards and a good quality transmitter, this should not be needed.

Step 4: More configuration options

Bitrate
You can change the bitrate to a higher value. (At this writing, values lower than 320 may be attempted but are not recommended or supported). If you are using 'L/R Asymmetry' in the FM settings, we recommend a higher bitrate than the standard 320.

Signal loss handling
If your connection to the decoder is not fully reliable and drops packets sometimes, you can enable forward error correction and change some encoder settings to reduce the impact of packet loss. Please note that, while forward error correction is pretty powerful, it cannot perform miracles - if your connection drops out for a longer period of time you will get a signal loss. The bottom line is you need a robust IP connection with predictable bandwidth.

Keyframe Interval: Reducing the effect of packet loss
By default, every piece of audio in μMPX needs the previous audio to be decoded. So, if a packet is lost, the next packet won't be decoded either. By default, every 5 seconds a packet with "reset data" is sent, which contains all the info needed to generate audio again. So, depending on when an outage occurs, a gap caused by a lost packet can last anywhere between a few milliseconds (the size of one packet) up to 5 seconds.

To manage trade-offs, you can lower the "Keyframe interval" slider. This will increase the rate that reset data is sent. At the lowest setting, every packet contains reset data, adding some overhead. The trade-off to more reset packets is less data is available for the encoded audio. Use this strategy only if your connection is reliable and wide enough. If your bitrate is not high enough to support the added overhead, you may be better off tolerating longer reset packet intervals.

Error Correction: Recovering lost packets

Omnia μMPX supports forward error correction. The theory of this system is that extra bandwidth is used to send redundant information so that some packet loss can occur without under running buffers. For example, if for every 100 packets sent, you also send 10 extra recovery packets (requiring 10% more bandwidth), if you lose no more than 10 of the total 110 packets, all the data can be reconstructed on the receiving end, insuring a complete signal gets received

In our example, the 10 recovery packets are generated immediately for every 100 packets that are sent, and - depending on your network - it could be that sending them all at once to the other side will fail and the network will drop some of the packets. Because of that, the "Rate limiter" will throttle the speed at which packets are sent (both for the normal and the recovery packets). Omnia SST automatically calculates the minimum bandwidth needed for both normal and recovery packets. Recovery packets are slightly bigger than normal packets so it's beneficial to allow some headroom.   In our example, settings of 360 kbit/s allow for a 320 kbit/s audio stream with decent forward error correction.

Please note that for these settings, at least 100 packets must have arrived on the other end before reconstruction can start. So if the first packet is dropped, it will take time before it can be reconstructed. The decoder must have enough buffer latency for it to still be able to use the recovered packet - if the audio at that point should already have been played, you'll still have a gap in your audio. The current version of the software sends 94 packets per second, so for every 94 packets or so you need to add a second of latency. You can set the latency (in seconds) for the decoder with option -d. So, in the example above with 10 recovery packets per 100 normal packets, set the latency to at least about 1.5 seconds (with -d 1.5, see below). The extra half second is there to reduce network jitter. Decoder default latency is 1 second. Generally, the higher the latency and buffering you use, the fewer problems from dropped packets. The inherent tradeoff of more error correction is greater signal delay.

One final consideration for latency vs. bandwidth. Packet loss tends to occur in bursts. So, the same number of recovery packets often provides the same "ballpark" amount of recovery, regardless of the number of packets over which they are calculated. So, 32/4 and 64/8 will give you the same network bandwidth, 64/4 gives you less network bandwidth with almost the same quality of recovery. But, you need to allow more time on the decoder end for the data to be recovered. 64/8 has a much smaller chance of losing audio, but the tradeoff is incurring more latency.

Latency

The default latency is set to 1 second. If you don't use error correction at all, you can attempt to set it lower. We have seen good results with values down to 0.2 seconds (example below), but results depend on many variables outside the control of SST, most importantly, your network topology and the quality of the hardware which you run the encoder and decoder on.

Backup file

If the connection drops, the µMPX decoder can switch to playing a backup audio file that you can provide.

--backup-file /path/to/file.wav points to the file (which must contain 192 kHz MPX data, in WAV or FLAC format). You will need to place this file in the same directory as the decoder.

--backup-timeout is the time in seconds of dropped signal before the backup file is triggered.

If multiple dropouts occur, the file will keep playing in a loop, picking up where it left off after the last dropout.

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UPDATES

Omnia SST software is routinely updated with new versions to add features and fix bugs. Check the Telos Alliance web site for the latest. We encourage you to sign-up for the email notification service offered on the site.

FEEDBACK

We welcome feedback on any aspect of Omnia SST or this manual. In the past, many good ideas from users have made their way into software revisions or new products. Please contact us with your comments.

TECHNICAL SUPPORT

You may reach our 24/7 support team around the clock by calling +1-216-622-0247 or by email at support@telosalliance.com. A variety of useful information is available on the Telos Alliance website (www.telosalliance.com/omnia).

ADDITIONAL CONTACT INFORMATION AND SUPPORT

For billing or non-emergency technical questions, call +1-216-241-7225 between 9:30am and 6:00pm USA Eastern time, Monday – Friday.

Our fax number is +1-216-241-4103.

General questions can be sent by email to inquiry@telosalliance.com.

Our mailing address is The Telos Alliance, 1241 Superior Avenue E., Cleveland, OH, 44114, U.S.A.

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