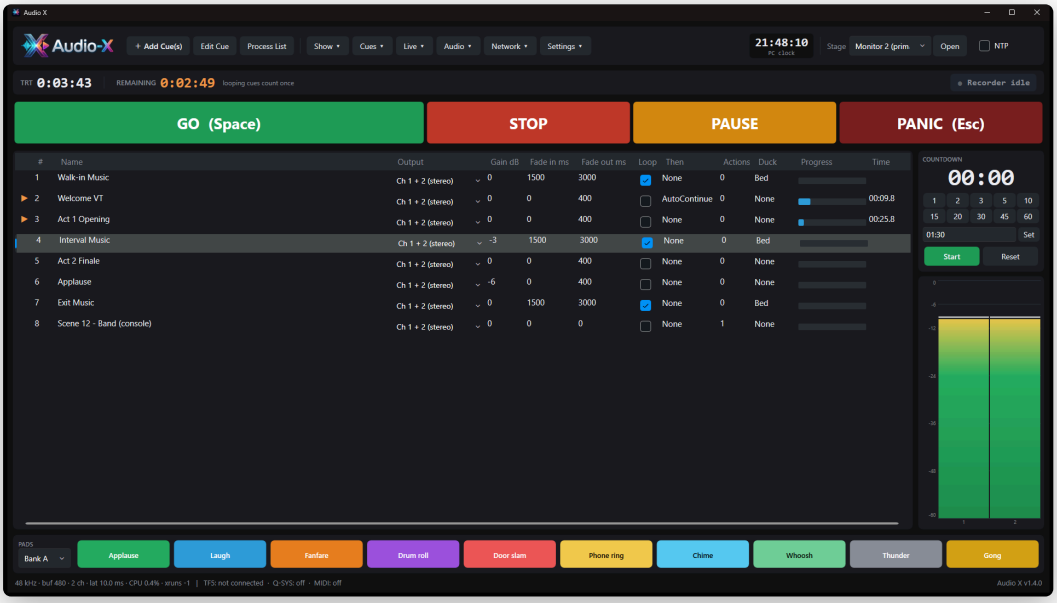
Audio X

User Manual



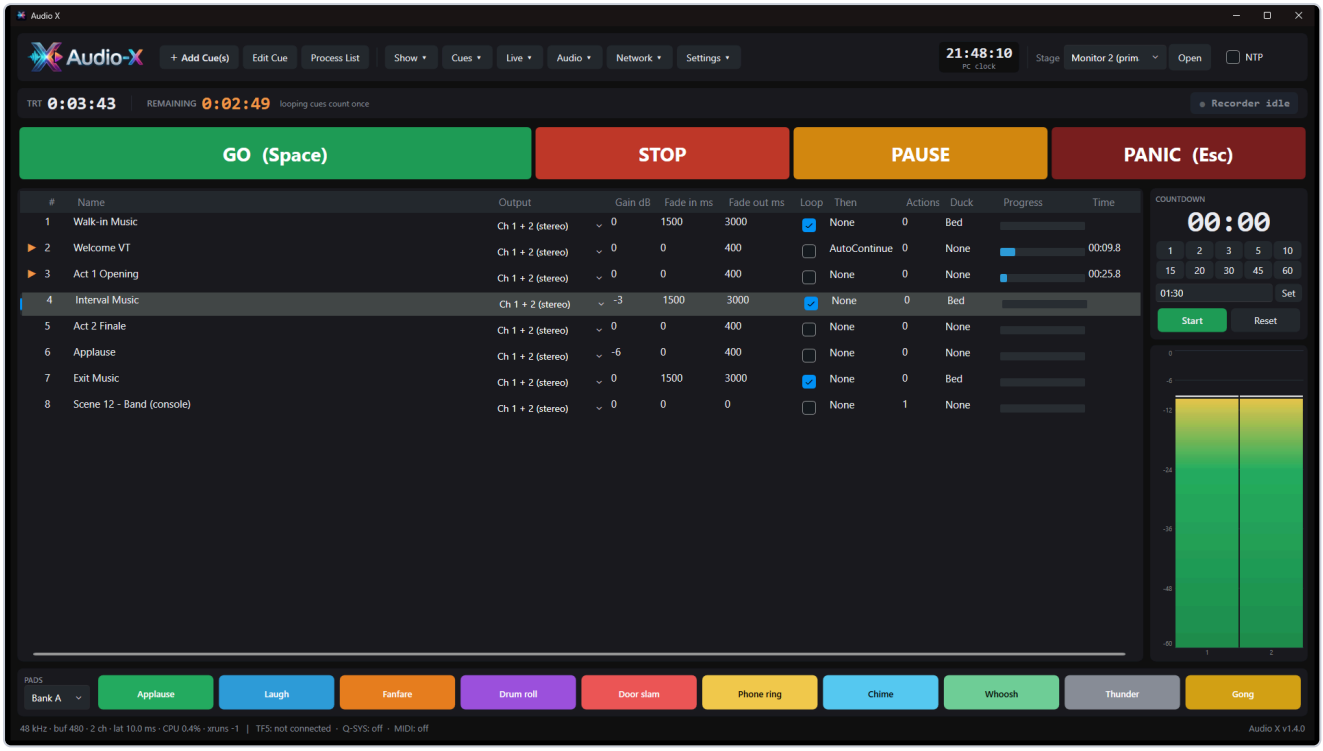
Version 1.4.0 · 4 November 2026

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audiox.info · contact@audiox.info

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Audio X User Manual



Audio X is a Windows program for playing audio in live venues. It runs a cue list for a show, an automatic radio station with idents, a soundboard, a talkback microphone, a multitrack recorder, a real-time analyser and sound level meter, and it controls other equipment (Yamaha TF consoles, MIDI devices and Q-SYS) from the same cue list. You can watch and control it from phones, tablets and other PCs on the same network.

This manual covers version **1.4.0**.

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1. Show Control: Yamaha, MIDI and Q-SYS - cue actions
2. Remote Control and the Phone App - web page and Android app
3. Stage Display - confidence monitor on other screens
4. Audio Tools - tone generator, surround up-mix, OpenDaw

Reference

1. Settings Reference
2. Menu Reference
3. Troubleshooting
4. Licence and Copyright - AGPL-3.0, warranty, source code

What you need

- Windows 10 or 11, 64-bit.
- An audio device: the PC's sound card (Windows Audio), a USB interface, or an ASIO device such as Dante Virtual Soundcard or the Yamaha Steinberg USB driver.
- FFmpeg, for the Media Converter and Process List (Audio X looks for it automatically; see Media Converter).
- Optional: a Yamaha TF console, a MIDI device or a Q-SYS Core on the network.

Getting Started

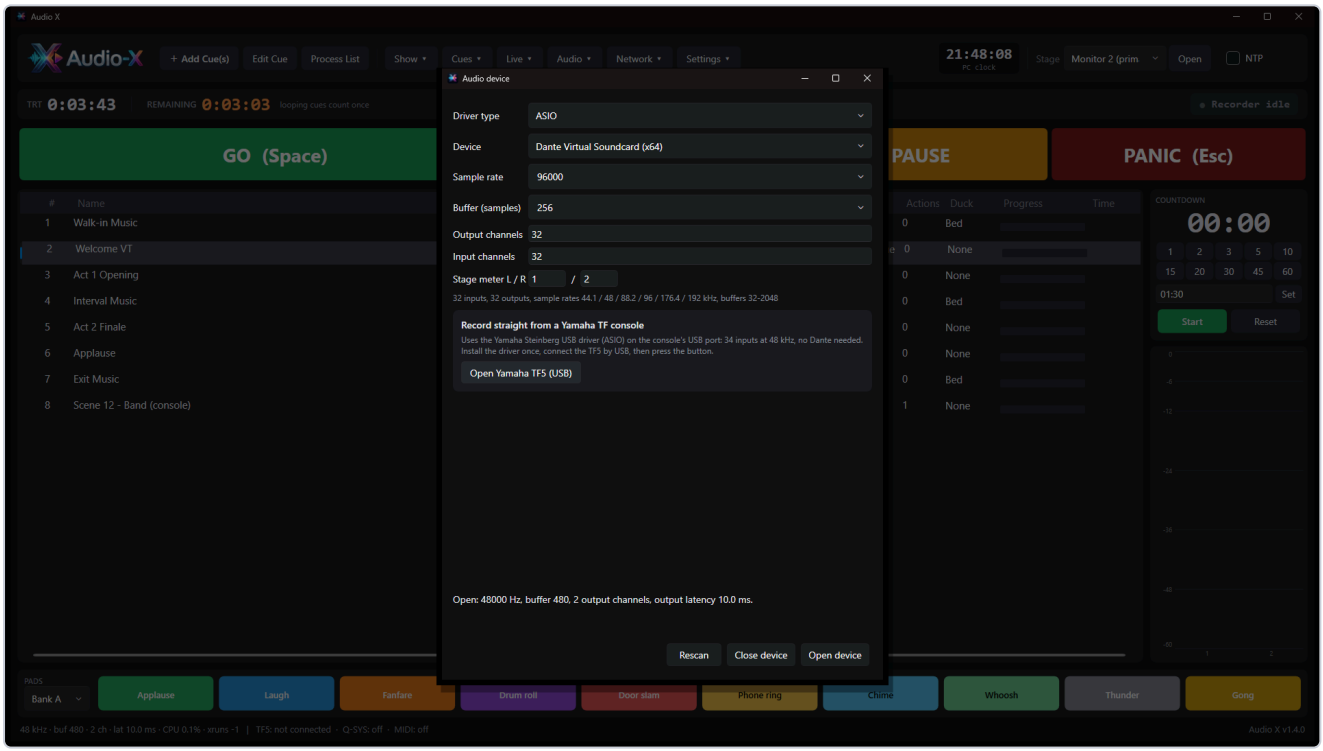
Install

Run `AudioX-Setup-1.4.0.exe` and follow the steps. A portable zip (`AudioX-1.4.0-win-x64.zip`) is also supplied: unzip it anywhere and run `AudioX.exe`. Your settings are kept in `%AppData%\AudioX` and your shows are `.axshow` files wherever you save them.

The first time Audio X listens on the network (for the remote control or the stage display), Windows asks whether to allow it. Choose **Private networks**.

1. Open the audio device

Audio X needs an audio device before it can play. Open **Settings > Audio device**.



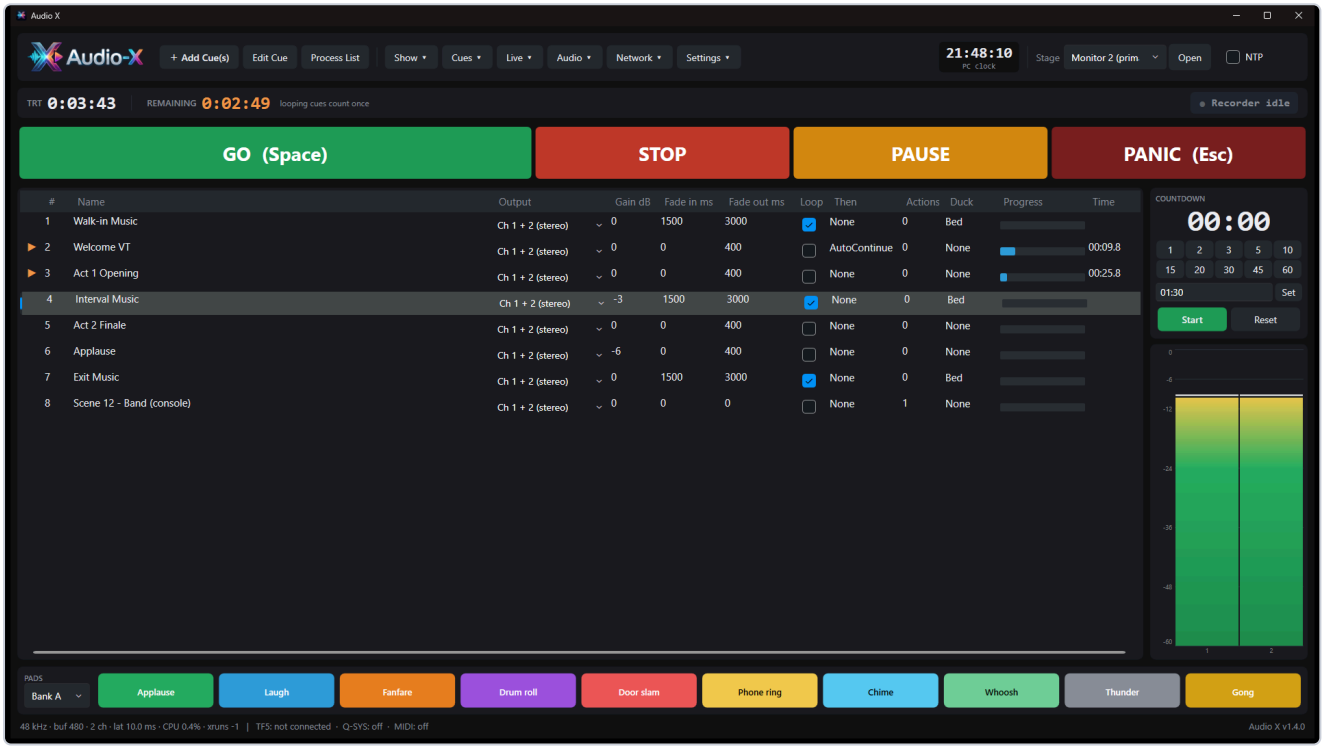
1. **Driver type**: *Windows Audio* for a normal sound card, *ASIO* for Dante Virtual Soundcard, the Yamaha Steinberg USB driver and most pro interfaces.
2. **Device**: pick the device. A line under the list tells you how many inputs and outputs it offers and which sample rates and buffer sizes it supports.
3. **Sample rate**, **Buffer (samples)**: 48000 Hz and a buffer of 128-480 samples are good starting points. A smaller buffer is quicker but needs a faster PC.
4. **Output channels**: how many outputs to open. **Input channels** (ASIO only): how many inputs to open for the recorder, the input monitor and the main-device microphone.
5. **Stage meter L / R**: which two outputs the stage display's stereo meter shows.
6. Press **Open device**. The line at the bottom shows the result (rate, buffer, channels, latency).

In **Preferences** you can tick *Open the audio device automatically* so Audio X opens it every time it starts.

For a Yamaha TF console connected by USB, press **Open Yamaha TF5 (USB)**: it opens the driver with all 34 channels. See Input Monitor and Recorder. For Dante, see Troubleshooting.

2. Make your first cue list

1. Press **+ Add Cue(s)** (top left) and choose audio files. Each file becomes a cue.
2. The first cue is on *standby*. Press **GO** (or the space bar) to play it. The next cue becomes the standby cue.
3. **STOP** fades everything out. **PAUSE** pauses and resumes. **PANIC** (Esc) stops everything at once.
4. Save with **Show > Save show**. Next time use **Show > Open show**, or tick *Open the last show automatically* in Preferences.



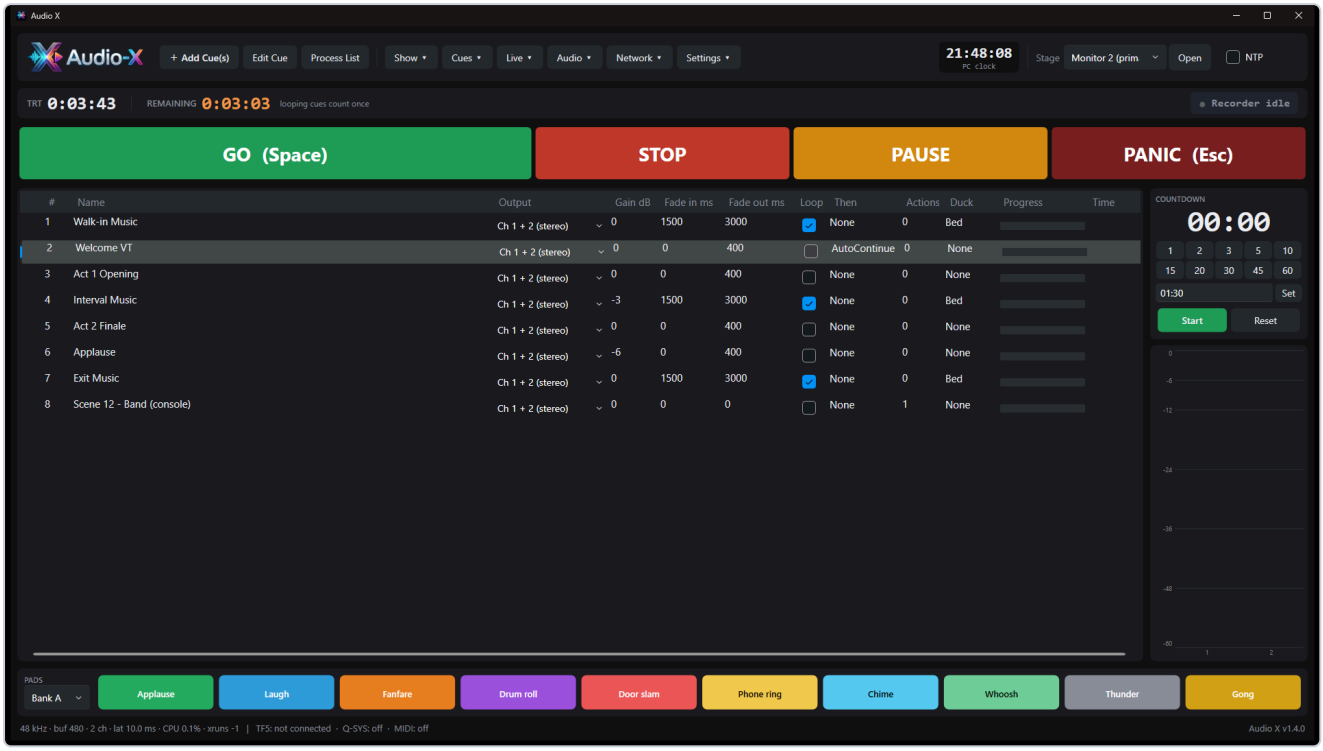
3. Choose where cues play

Every cue has an **Output** drop-down. The default is *Ch 1 + 2 (stereo)*. A mono file plays to both channels of a pair. You can send a cue to another pair, a single channel, or all outputs.

Next steps

- Get the whole list ready quickly: Process List and Normalise.
- Run music between the show: Radio.
- Add effects for instant playback: Soundboard.
- Control it from a phone: Remote Control and the Phone App.

The Main Window



The top bar

Item	What it does
+ Add Cue(s)	Adds audio files to the cue list, one cue per file.
Edit Cue	Opens the Cue Editor for the selected cue.
Process List	Converts every cue to MP3, trims silence and normalises the loudness. See Process List and Normalise.
Show, Cues, Live, Audio, Network, Settings	The six menus. See the Menu Reference.
Clock	The show time (the PC clock, corrected by the NTP time server when it is on). Underneath: the SMPTE timecode being sent or the time source. Click it for the timecode settings.
Stage / Open	Choose a screen and press **Open** to show the stage display on it.
NTP	Tick to keep the clock in step with a network time server.

Running time

TRT is the total running time of all cues. **REMAINING** is what is playing now plus every cue from the standby cue on. Looping cues count once.

The transport buttons

Button	Key	What it does
GO	Space	Plays the standby cue and moves standby to the next cue.
STOP		Fades out everything that is playing.
PAUSE		Pauses; press again to resume.
PANIC	Esc	Stops cues, the tone generator and the soundboard at once. The radio and a recording keep running.

All keys can be changed in **Settings > Key mapping**. F11 switches full screen on and off.

The cue list

Column	Meaning
# and **Name**	The cue number and name. Double-click to edit. An orange arrow marks a cue that is playing; a blue bar marks the standby cue.
Output	Where the cue plays.
Gain dB	Level of the cue. Normalise and Process List set it for you.
Fade in / out ms	Fade times.
Loop	Repeats the cue until stopped.
Then	*None*: nothing. *AutoContinue*: when this cue starts, the next cue starts too. *AutoFollow*: the next cue starts when this one finishes.
Actions	How many show-control actions the cue fires.
Duck	*Bed* (music that the radio ducker turns down), *Trigger* (an announcement that turns beds down), or *None*.
Progress, Time	How far a playing cue has got and its time left.

The right-hand column

- **Countdown timer**: presets of 1 to 60 minutes, or type a time (`5` , `1:30` , `1:05:00` , `90s` , `2.5m` , `1h`) and press **Set**. **Start** starts, pauses and resumes; **Reset** clears. It flashes orange in the last 10 seconds and red at zero, and shows on the stage display and the web page.
- **Output meters**: one bar per open output channel, 0 to -60 dBFS.

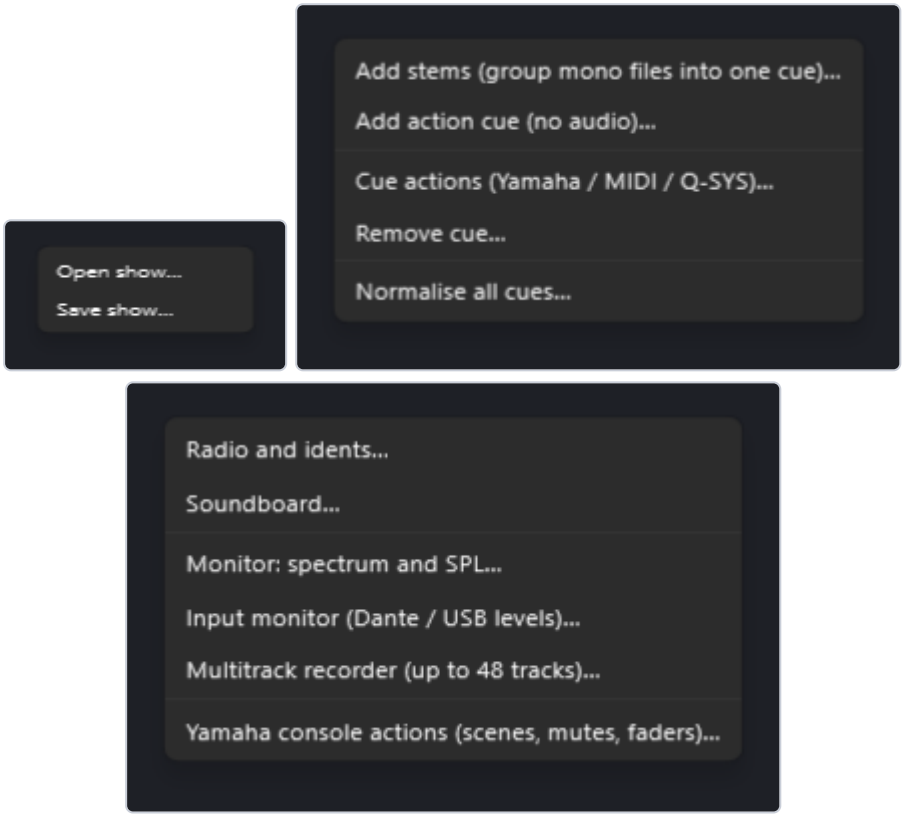
The pad row

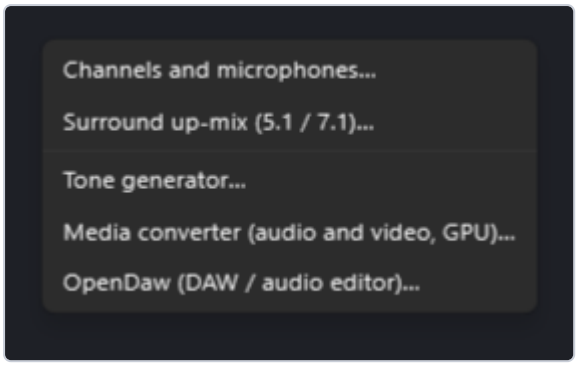
The ten pads at the bottom are the first ten pads of the soundboard bank chosen in the little *Bank* drop-down. Click a pad to play it, right-click to edit it, click an empty pad to assign a sound. See Soundboard.

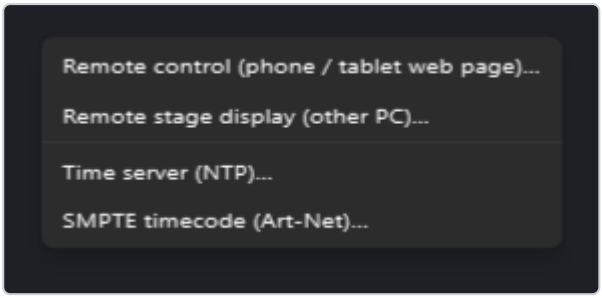
The status bar

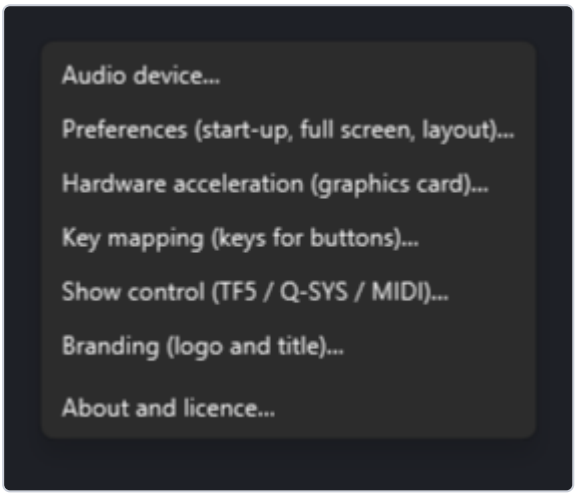
Shows the audio device (rate, buffer, channels, latency, CPU), the link state of the Yamaha console, Q-SYS and MIDI, and the live sound level when the analyser is running (it says OVER LIMIT when the limit is exceeded). The version is at the bottom right.

The menus




Channels and microphones...
Surround up-mix (5.1 / 7.1)...
Tone generator...
Media converter (audio and video, GPU)...
OpenDaw (DAW / audio editor)...


Remote control (phone / tablet web page)...
Remote stage display (other PC)...
Time server (NTP)...
SMPTE timecode (Art-Net)...


Audio device...
Preferences (start-up, full screen, layout)...
Hardware acceleration (graphics card)...
Key mapping (keys for buttons)...
Show control (TF5 / Q-SYS / MIDI)...
Branding (logo and title)...
About and licence...

Cues and Shows

A **show** is a cue list saved as a `.axshow` file. A **cue** plays one audio file (or a group of files), or does something to other equipment without playing audio.

Adding cues

- **+ Add Cue(s)** adds one cue per file you choose.
- **Cues > Add stems** groups several mono files into one cue (for multitrack stems that must start together).
- **Cues > Add action cue** adds a cue with no audio, only actions such as recalling a scene on the Yamaha console. The demo list in the pictures ends with one: *Scene 12 - Band (console)*.
- **Cues > Remove cue** deletes the selected cue.

Selecting and firing

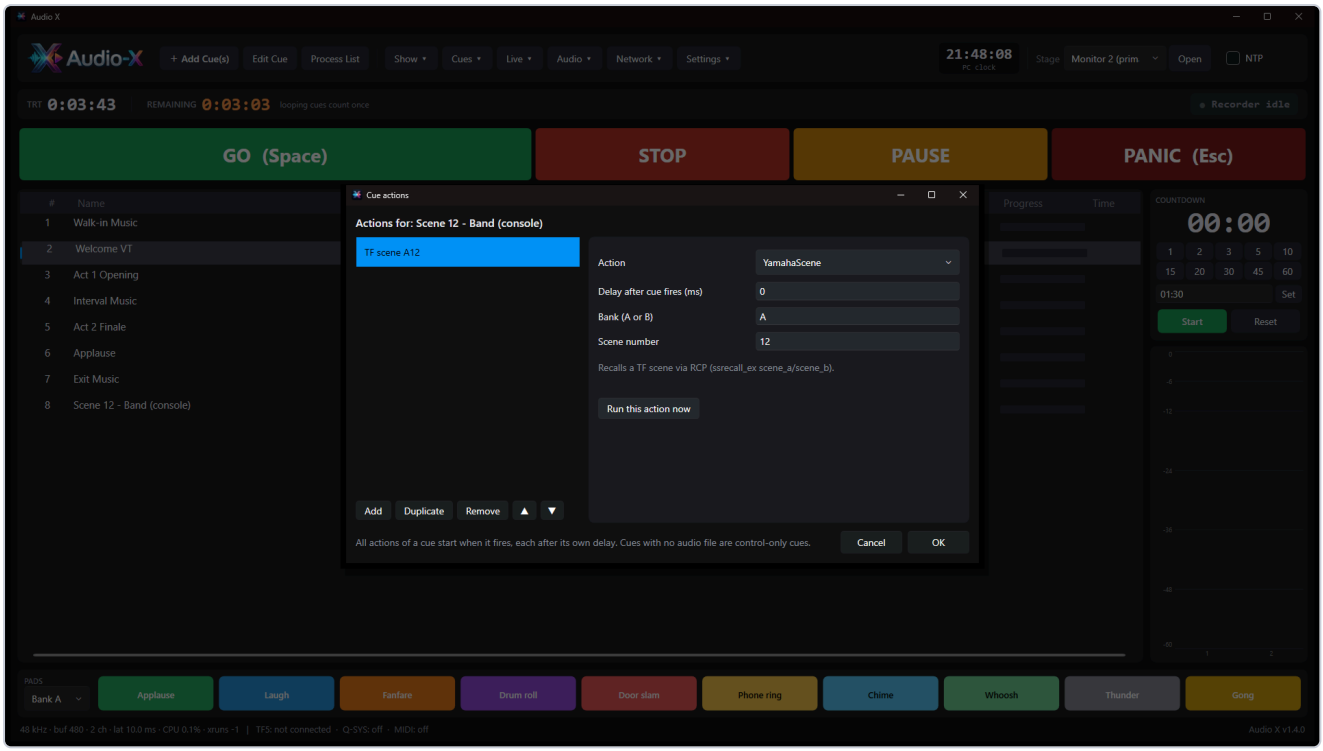
- Click a cue to put it on **standby**. **GO** fires the standby cue.
- Cues can start together: set **Then** to *AutoContinue* on a cue and the next cue starts at the same moment; set it to *AutoFollow* and the next cue starts when this one ends.
- A **Loop** cue repeats until you press STOP.

Per-cue settings

Each column of the cue list is a setting: output, gain, fades, loop, then, duck. For trims and waveform work use the Cue Editor.

Cue actions

Cues > Cue actions opens the actions of the selected cue.



Choose an action type (Yamaha scene, fader or mute, raw RCP line, MIDI program / control / note / SysEx, Q-SYS control or snapshot), fill in its values and set a **delay** in milliseconds. All actions of a cue start when the cue fires. See Show Control.

Normalising and processing

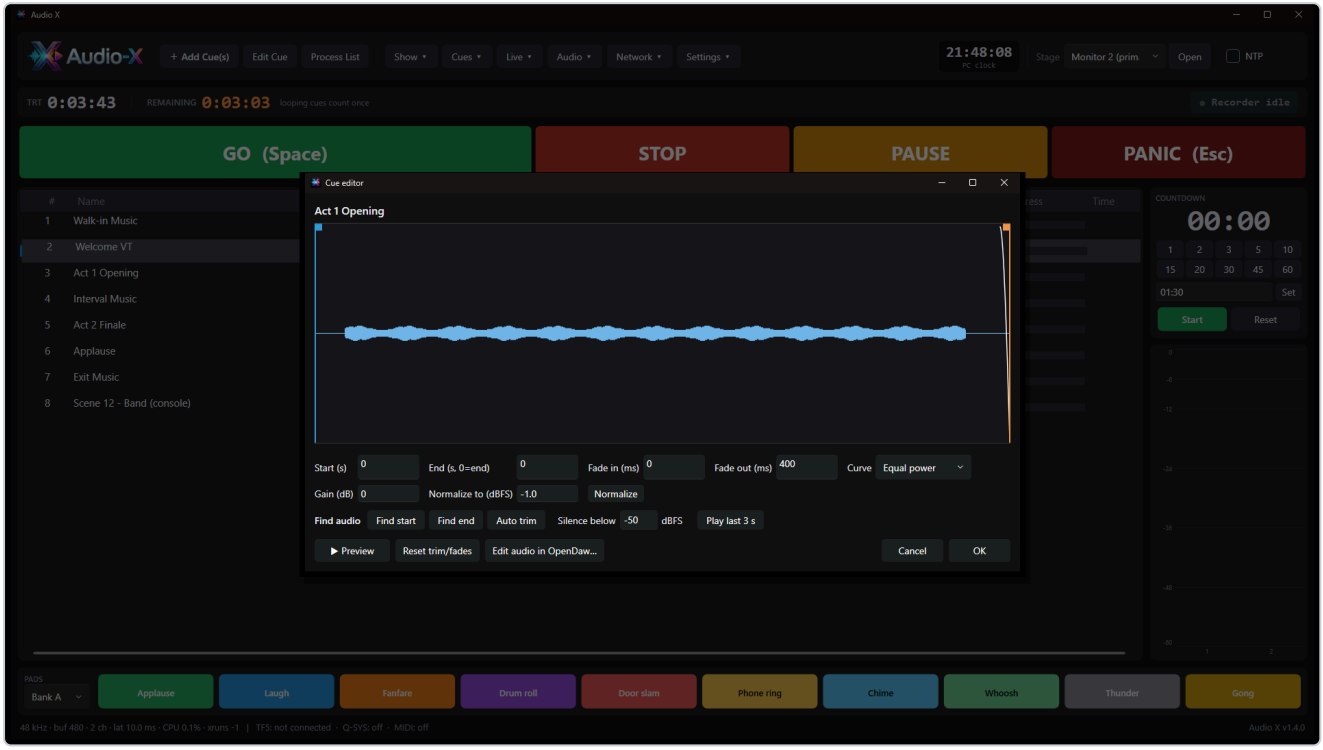
- **Cues > Normalise all cues** measures the loudness of the cue list (and optionally the radio songs and idents) and sets a gain for each. See Process List and Normalise.
- **Process List** (top bar) does conversion, trimming and normalising in one go.

Saving

- **Show > Save show** writes the cue list to a `.axshow` file. Audio files are referenced, not copied: keep them where Audio X can find them.
- **Show > Open show** loads one. Tick *Open the last show automatically* in Preferences to load the last show at start-up.

Cue Editor

Select a cue and press **Edit Cue**. The editor never changes your audio file: it stores trim, fades and gain in the cue.



The waveform

The waveform shows the whole file. Drag the **start** and **end** markers to trim, or type the times. Anything before the start marker or after the end marker is not played.

Auto trim

- **Find start** moves the start marker to where the audio really begins.
- **Find end** moves the end marker to where it really ends.
- **Auto trim** does both.
- **Silence** sets what counts as silence in dBFS (default -50). Raise it (for example -40) for tracks with a noisy start or end; lower it (for example -60) to keep quiet fades.
- **Play end** previews the last three seconds before the end marker, so you can hear the cut.

Audio X leaves a tiny margin (10 ms before the first sound, 50 ms after the last) so nothing is clipped.

Fades and gain

Set **fade in** and **fade out** in milliseconds and choose the fade curve. **Gain** is the cue's level in dB; **Normalise** sets it so the trimmed part reaches a target loudness.

Preview

Preview plays the cue through the audio engine with the trims and fades applied.

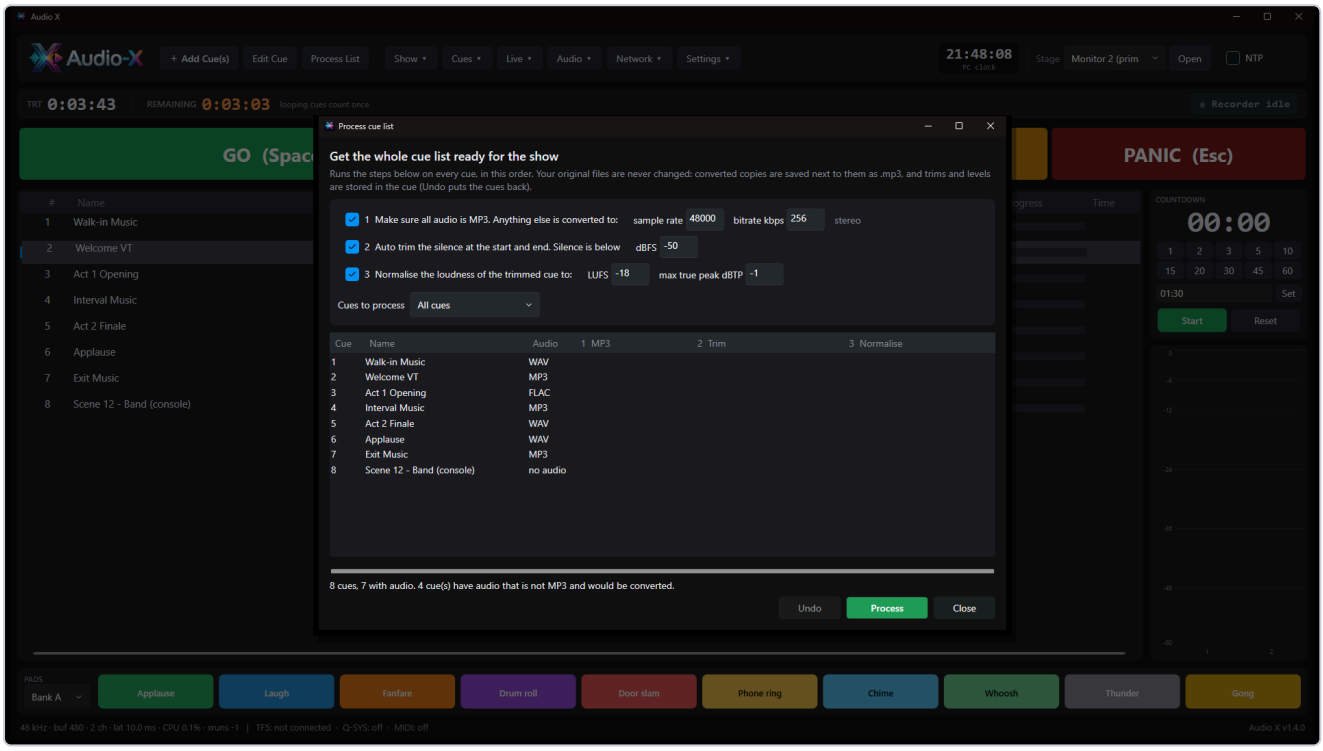
Edit audio in OpenDaw

For real audio editing (cutting, effects, mixing) press **Edit audio in OpenDaw**. Audio X opens the file in OpenDaw, a separate free program, and reloads the waveform when you come back. See Audio Tools.

Process List and Normalise

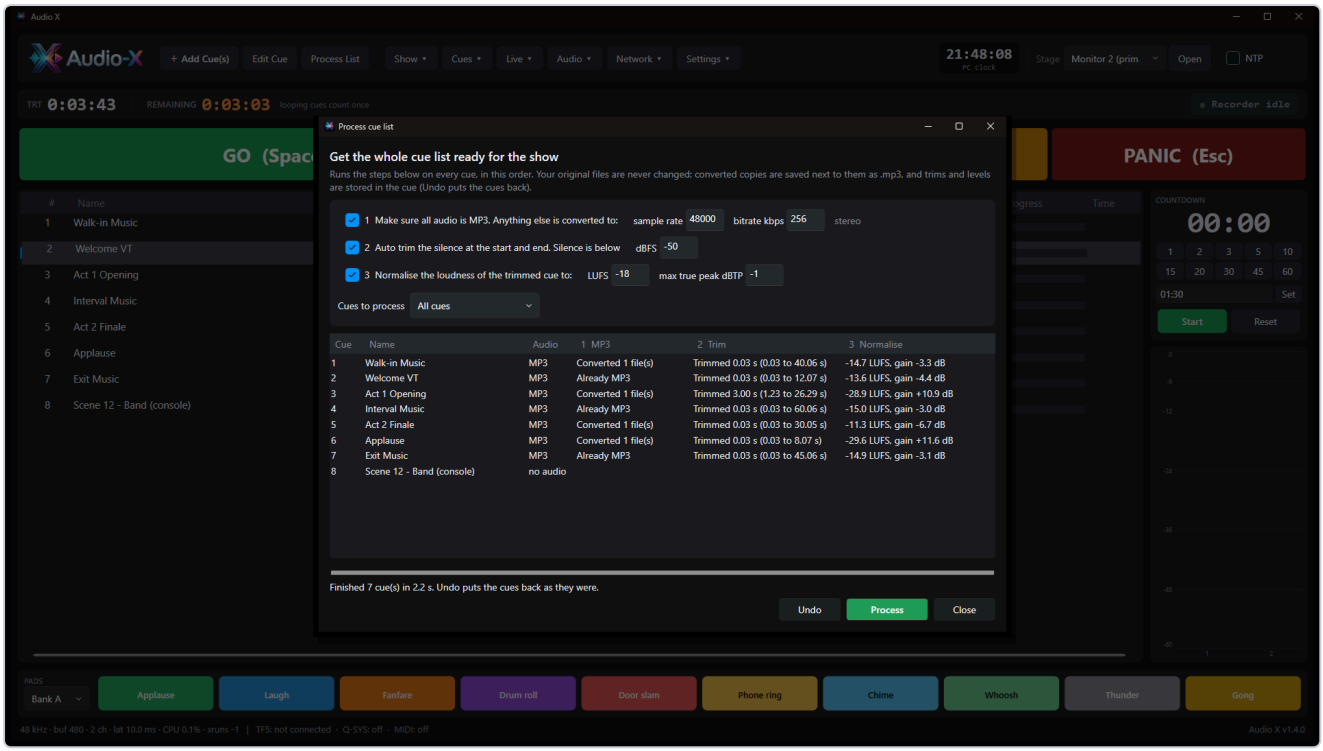
Process List

The **Process List** button (top bar) gets a whole cue list ready for a show. It runs three steps on every cue, in this order, so the loudness is measured on the finished, trimmed file.



Step	What it does
1 Make sure all audio is MP3	Any file that is not an MP3 is converted to MP3 at **48000 Hz, 256 kbps, stereo** (you can change the values). Files that are already MP3 are left alone. A file used by two cues is converted once.
2 Auto trim	Moves the start and end of each cue to the first and last sound, using the silence threshold (default -50 dBFS) and the same margins as the Cue Editor.
3 Normalise	Sets each cue's gain so the trimmed cue plays at **-18 LUFS**, with the true peak kept below **-1 dBTP** (the same standard values as Normalise all cues).

Tick or untick each step, change the values, and choose **All cues** or **Selected cue only**. Then press **Process**.



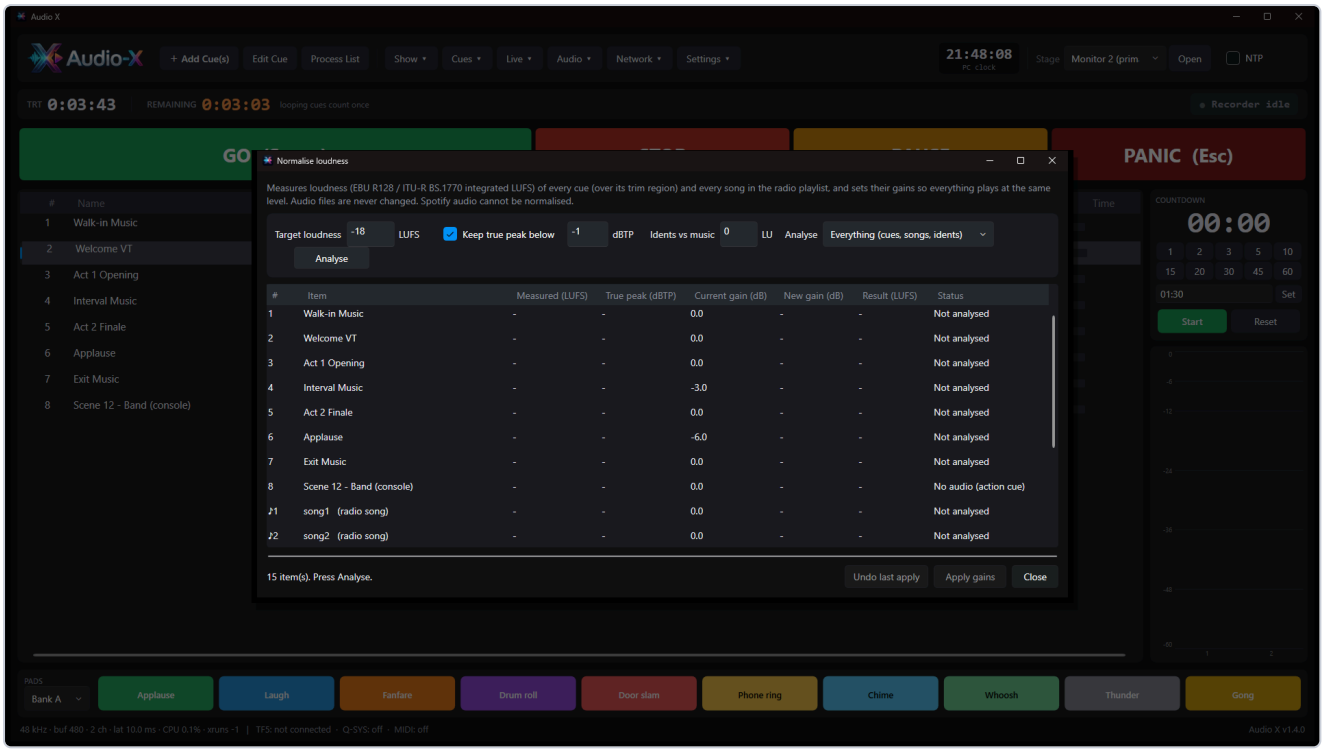
The table shows what happened to each cue. In this example the cue *Act 1 Opening* had 1.2 s of silence at the start and 1.8 s at the end, which were trimmed, and its quiet recording was lifted by 10.9 dB. A cue with no audio (an action cue) is skipped.

Your original files are never changed. Converted copies are saved next to the originals with the extension `.mp3`; the cue list is pointed at the copies. Trims and gains are stored in the cues. **Undo** puts every cue back as it was (the converted MP3 files stay on disk).

Process List needs FFmpeg to convert (see Media Converter). Trimming and normalising need no FFmpeg. Several files convert at the same time; the number is set in Hardware Acceleration.

Normalise all cues

Cues > Normalise all cues measures loudness (ITU-R BS.1770, integrated LUFS and true peak) without changing any files.

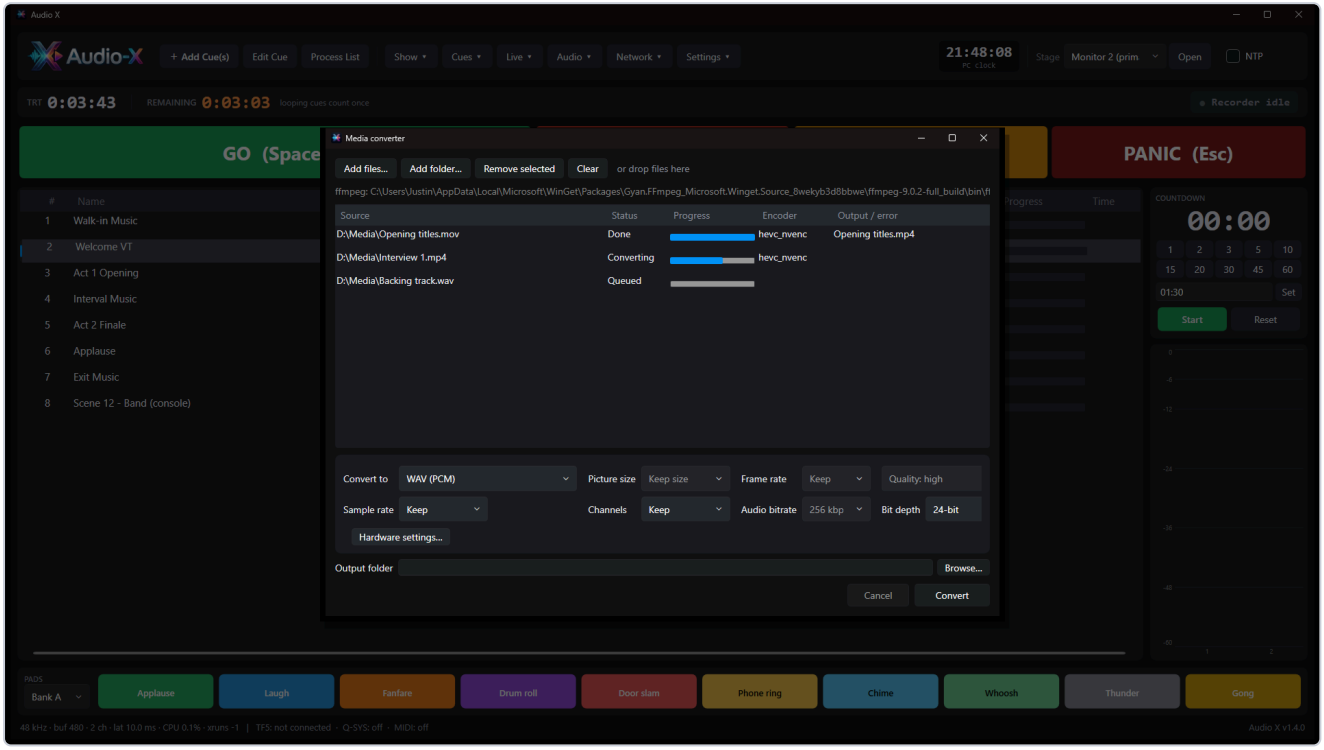


- Choose the **scope**: everything (cues, radio songs, radio idents), cues only, songs only, idents only, or the selected cue.
- Set the **target** LUFS (default -18) and the **true-peak ceiling**. A cue that would exceed the ceiling gets less gain and is marked *peak-limited*.
- Press **Analyse**, check the table (measured level, true peak, current gain, new gain, result), then **Apply**. **Undo** restores the old gains.
- Idents are normalised to match the radio music's playing level, plus an optional offset in LU.

Use Normalise when you only want levels. Use Process List when you want conversion, trimming and levels together.

Media Converter

Audio > Media converter converts audio and video files to other formats. It scales pictures up or down, changes codecs and containers, and uses the graphics card to encode video when it can.



Converting files

1. Add files with **Add files**, **Add folder** (every media file in the folder and its sub-folders), or drag them onto the window.
2. Choose **Convert to** (see the list below), then the other settings.
3. Choose an **Output folder**, or leave it empty to save next to each source file. Audio X never overwrites a file; it adds *(1)* to the name if needed.
4. Press **Convert**. Each file shows its progress, the encoder used and the output name. **Cancel** stops the job and removes unfinished files.

Several files are converted at the same time (see Hardware Acceleration). The summary line at the bottom gives the total time.

Formats

Group	Choices
Audio	WAV (16, 24 or 32-bit float), FLAC (16 or 24-bit), MP3, AAC (m4a), Ogg Vorbis, Opus
Video	MP4 H.264 + AAC, MP4 H.265 (HEVC) + AAC, MKV H.264 + AAC, MKV H.265 + AAC, MKV AV1 + Opus, WebM VP9 + Opus, MOV ProRes 422 HQ + PCM, MOV H.264 + AAC

Re-wrap	MP4 copy video + AAC: the picture is not re-encoded, so there is no loss and it is fast

A video source converted to an audio format keeps only the audio (the first audio track).

Up, down and across

- **Picture size** scales the picture up or down to 4K (2160p), 1440p, 1080p, 720p, 576p, 480p or 360p. The width follows, so the shape is kept.
- **Frame rate** keeps the source rate or sets 24, 25, 30, 50 or 60.
- **Quality** from *draft (small)* to *best (large)* sets the video quality for H.264, H.265, AV1 and VP9.
- **Sample rate**, **Channels**, **Audio bitrate** and **Bit depth** apply to the audio.
- Converting between codecs and containers (for example H.264 MP4 to H.265 MKV, or MOV to WebM) is just a choice of **Convert to**.

Using the graphics card

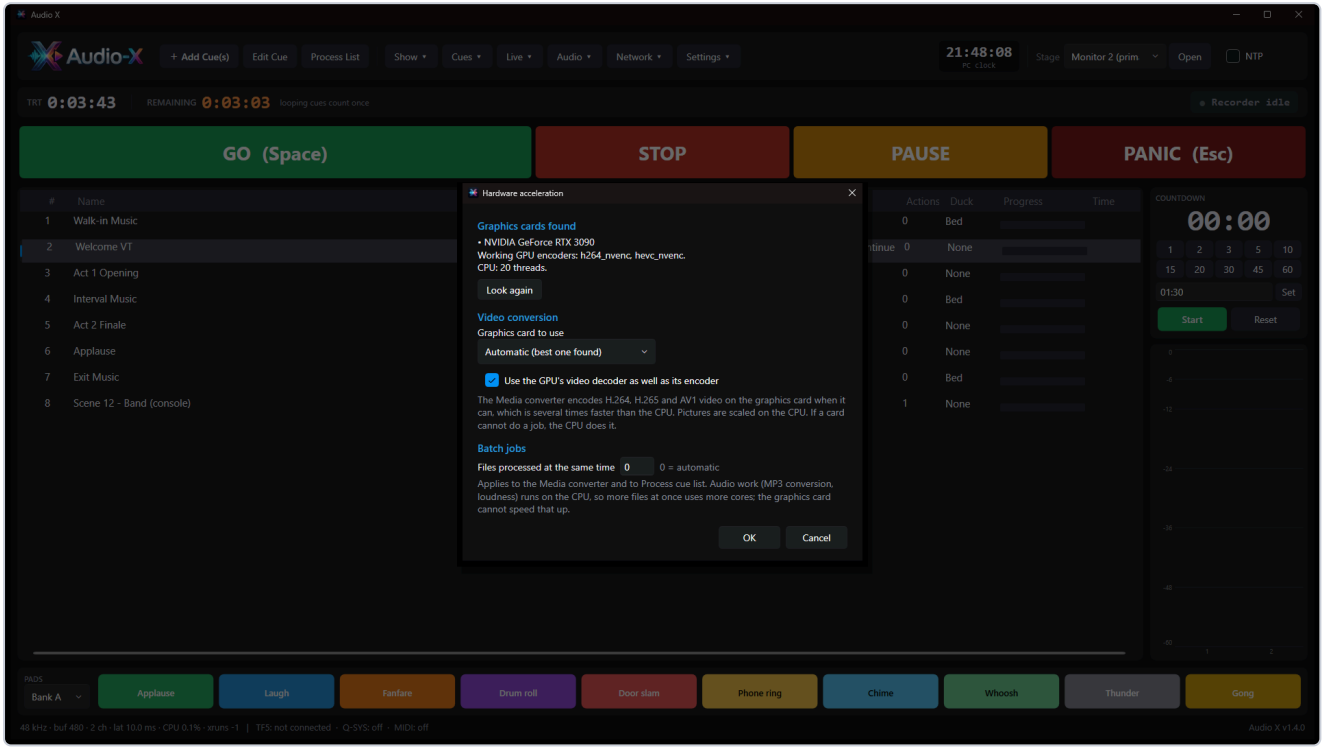
Tick **Use the graphics card (GPU) where possible** (it appears for video formats). H.264, H.265 and AV1 are encoded on the GPU when this PC has a card that can do it; the line next to the tick shows what was found, and the **Encoder** column shows what each file used (for example `hevc_nvenc`). If the card cannot do a job, the CPU does it. **Hardware settings** opens Hardware Acceleration.

FFmpeg

The converter uses FFmpeg. Audio X looks for it in the path you set, then the Audio X folder, then the system PATH, then the WinGet links folder. If it is not found, the window says so; install FFmpeg (for example `winget install Gyan.FFmpeg`) or press **Set ffmpeg** and choose `ffmpeg.exe`.

Hardware Acceleration

Settings > Hardware acceleration controls how Audio X uses the PC's graphics card and processor for batch work.



Graphics cards found

The window lists the graphics cards in the PC and the **GPU video encoders that really work**. Audio X test-runs each one on a tiny clip (NVIDIA NVENC, Intel Quick Sync, AMD AMF; H.264, H.265 and AV1) instead of trusting a list, so only encoders that start are offered. **Look again** repeats the check, for example after a driver update.

Video conversion

- **Graphics card to use**: *Automatic* (the best card found), *Off (CPU only)*, or a vendor: *NVIDIA (NVENC)*, *Intel (Quick Sync)*, *AMD (AMF)*.
- **Use the GPU's video decoder as well as its encoder** speeds up reading the source video.

The Media Converter then encodes H.264, H.265 and AV1 on the card. In Audio X's own tests a 20-second 1080p clip encoded to H.265 about **four times faster** on an NVIDIA RTX 3090 than on the CPU. Your speed-up depends on the card and the footage. Pictures are scaled on the CPU.

Batch jobs

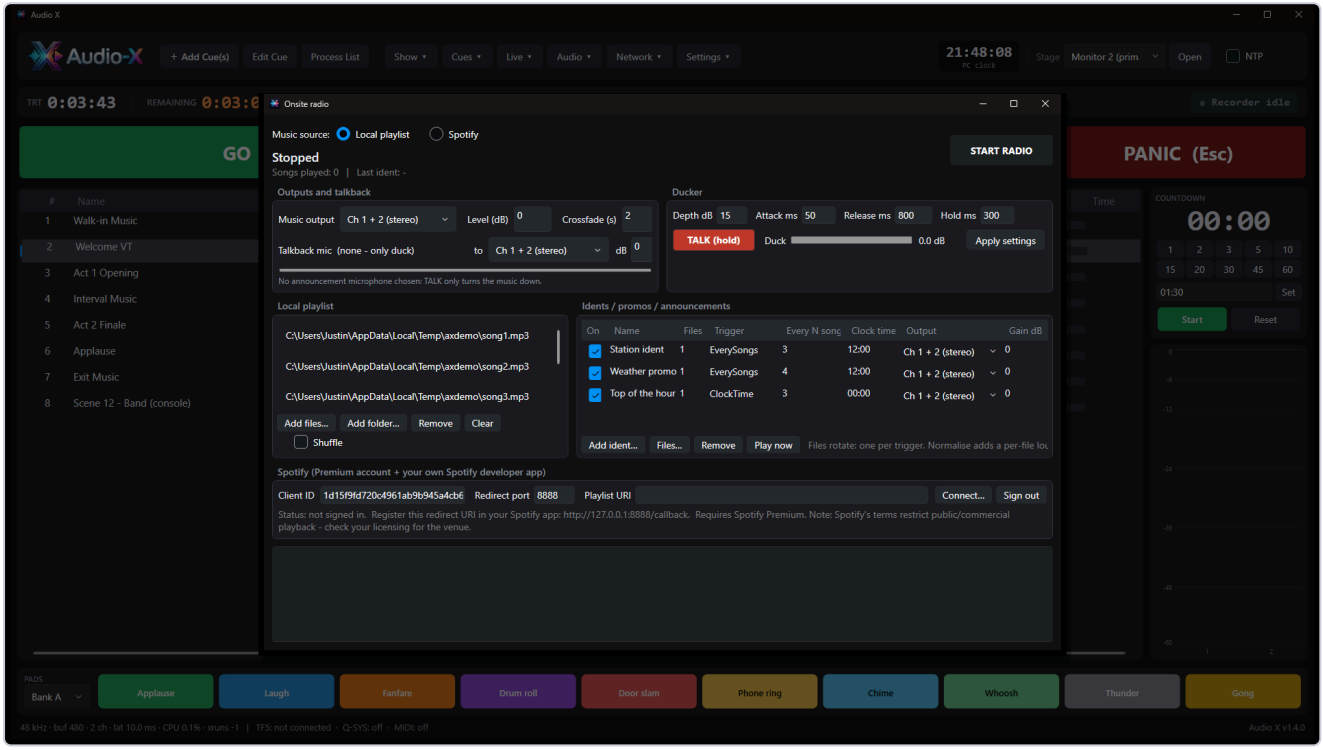
Files processed at the same time (0 = automatic) applies to the Media Converter and to Process List. Higher numbers use more processor cores.

What the GPU cannot do

MP3 conversion, trimming and loudness analysis are processor work with no graphics-card version. For these jobs the speed comes from processing several files in parallel. Audio playback, mixing and the microphone processing also run on the CPU, in Audio X's real-time audio engine.

Radio

Live > Radio and idents runs an automatic radio station: a playlist of songs with idents, promos and announcements mixed in between, with no gaps.



Starting

1. Choose the **music source**: *Local playlist* or *Spotify*.
2. Add songs (see below), then press **START RADIO**. The window shows what is playing, how many songs have been played and the last ident.
3. Press the button again to stop. The radio can also be started and stopped from the phone page and with a key, and it can start automatically when Audio X starts (Preferences).

Local playlist

- **Add files** and **Add folder** (every audio file in the folder and its sub-folders). You can also drag files and folders onto the list. **Remove** and **Clear** edit the list.
- **Shuffle** plays every song once, in random order, before any song repeats.
- Songs are loaded ahead of time and crossfaded, so there is never dead air. The crossfade time is set under *Outputs and talkback*.

Idents, promos and announcements

Idents are short sounds that play between songs.

Column	Meaning
On	Switch an ident on or off.
Name, **Files**	The name and how many audio files it has. Files rotate randomly, never the same one twice in a row. Press **Files** to choose them.
Trigger	*EverySongs*: after every N songs. *ClockTime*: at a time of day.
Every N songs / Clock time	The N, or the time (for example `00:00` for the top of the hour; it uses the NTP-corrected clock and plays once).
Output, **Gain dB**	Where the ident plays and its level.

- Idents play **one at a time, between two songs**, and are mixed over the end of the song before them. If two are due at once, they wait for the next gap, one per gap, in random order.
- **Play now** plays the selected ident straight away (the web page has ident buttons too). It is skipped if another ident is already playing.
- **Normalise** (see Process List and Normalise) can set matching levels for songs and idents.

Outputs and talkback

- **Music output**, **Level (dB)** and **Crossfade (s)** set where the music plays, how loud and how long the blend is.
- **Talkback mic**: shows the announcement microphone. Choose it in **Audio > Channels and microphones** (see Microphones and Talkback). **to** chooses the outputs where the announcement is heard, and **dB** its level.
- The level bar and the status line under it show the announcement microphone live, and tell you in plain words if something is wrong ("The audio device is not open", "Waiting for audio from the web device").

Ducker

When **TALK** is held (button, phone page or key) or an announcement cue plays, the music is turned down so the voice is clear, then comes back.

- **Depth dB** how far the music goes down, **Attack ms**, **Release ms**, **Hold ms**.
- **Duck** shows the current reduction. **Apply settings** saves the ducker and talkback settings.

Spotify

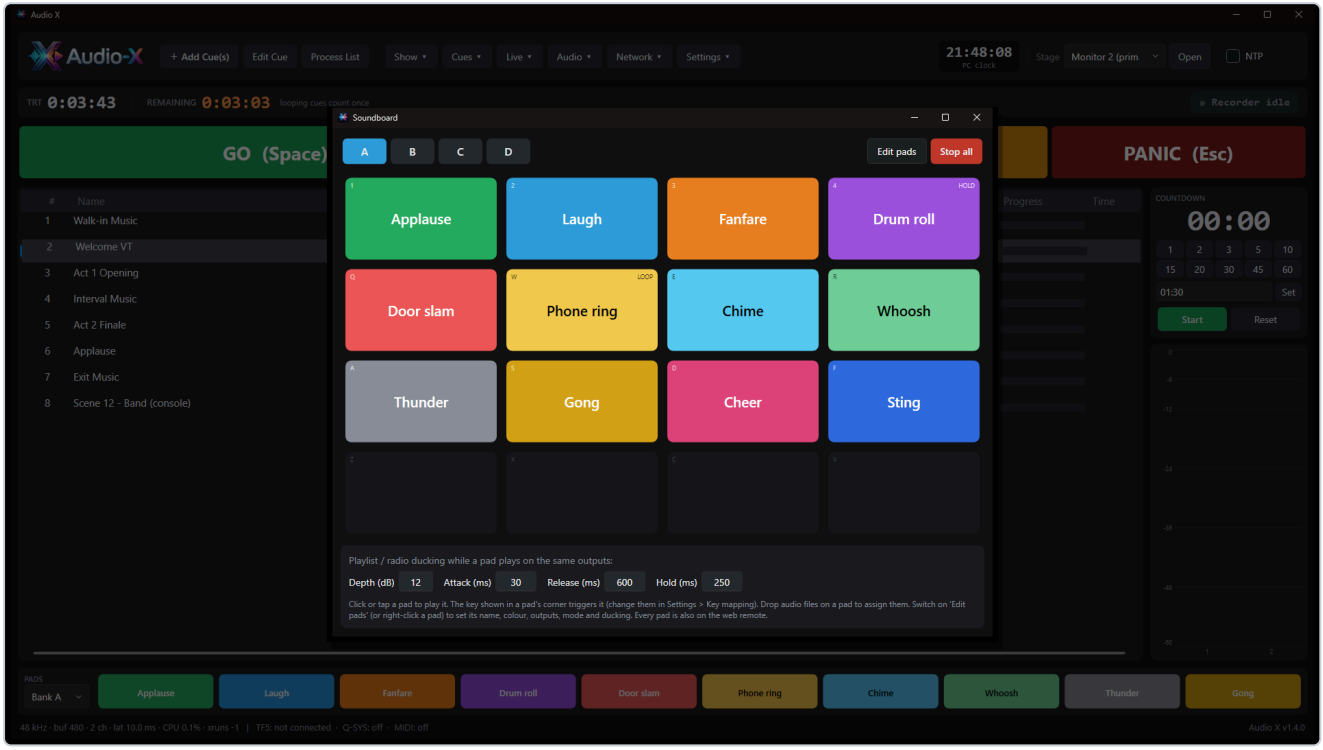
Spotify needs a **Premium** account and a free Spotify developer app.

1. On the Spotify developer dashboard create an app with *Web Playback SDK* and *Web API* ticked.
2. Add the redirect address shown in the Radio window (default `http://127.0.0.1:8888/callback`).
3. Paste the **Client ID** and a playlist or album address (`spotify:playlist:...`) into the window and press **Connect**. Your browser opens so you can sign in there.
4. Choose *Spotify* as the music source and press **START RADIO**.

Spotify cannot take an ident between songs, so idents play over the music (which is ducked), one after the other. Spotify's terms restrict public and commercial performance: check your venue licence, or use the local playlist.

Soundboard

Live > Soundboard has 4 banks (A to D) of 16 pads. A pad starts in about 25 ms because its sound is already loaded.



Playing

- Click or tap a pad to play it. The key shown in the corner of a pad also plays it (change keys in **Settings > Key mapping**).
- **Stop all** stops every pad. PANIC stops pads too.
- The first ten pads of the chosen bank are also in the row at the bottom of the main window, and every pad is on the phone page.

Setting up pads

Drop an audio file on a pad to assign it. Press **Edit pads** (or right-click a pad) to set:

- **Name** and **colour**
- **Outputs**: `1-2`, `3`, `1,2,5-6` or `all`
- **Mode**: *One shot* (press again to restart), *Toggle* (press to start, press to stop), *Hold* (plays while held), *Loop*
- **Gain** and **fades**
- **Duck the playlist and radio**

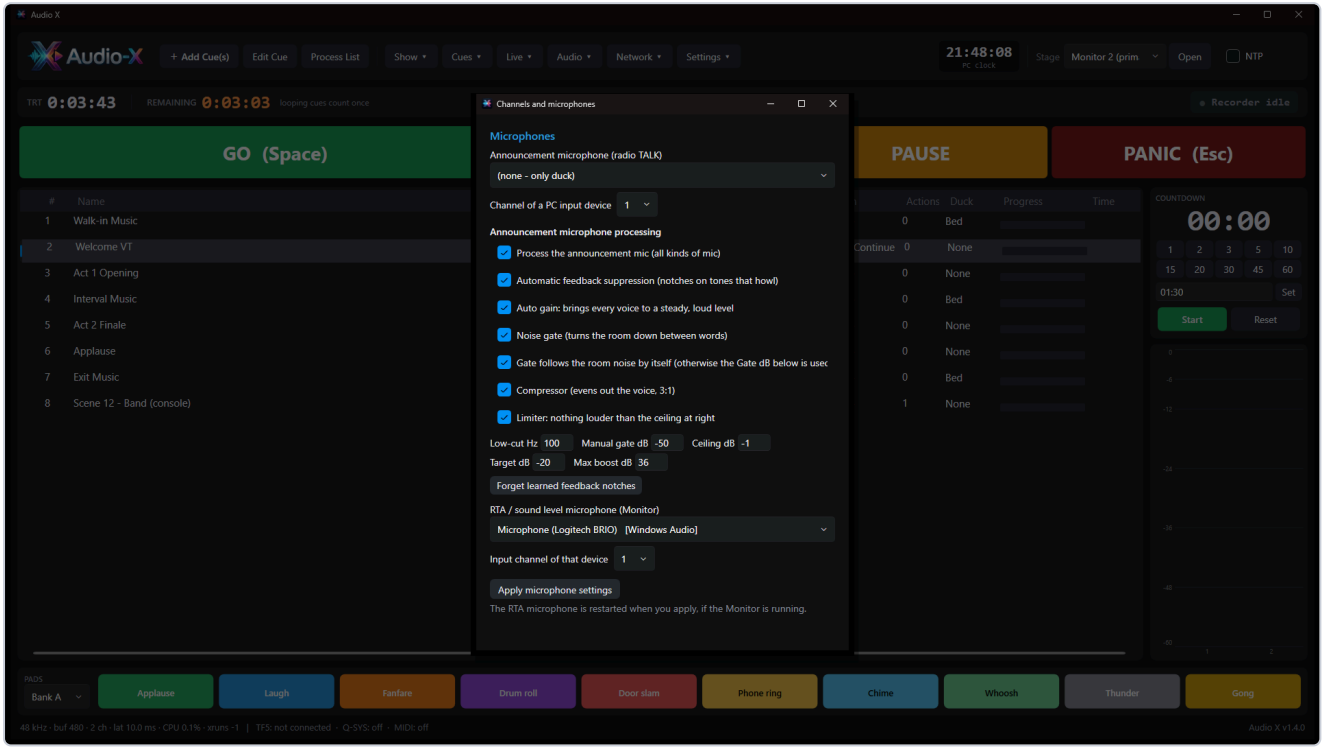
The corner of a pad shows HOLD or LOOP for those modes.

Ducking per output

While a pad that has *duck* ticked plays, the cues, radio and loopback audio on the **same outputs** is turned down, then recovers. The settings at the bottom of the window are **Depth** (default 12 dB), **Attack** (30 ms), **Hold** (250 ms) and **Release** (600 ms). Other outputs are not touched. A mono pad plays on every selected output; a stereo pad uses the first two.

Microphones and Talkback

Audio > Channels and microphones sets the two microphones Audio X can use. They are different devices, so each has its own choice.



Announcement microphone (radio TALK)

This is the microphone used for announcements. While **TALK** is held, it is heard on the outputs chosen in the Radio window and the radio music is turned down. The microphone can be:

- **Any hardware input of this PC**: a USB microphone, the built-in input or an interface. They appear as *PC input: name [Windows Audio]*. Use **Channel of a PC input device** for multi-channel devices.
- **An input of the main audio device**: for example the Dante or Yamaha USB channel the stage microphone is on.
- **The microphone of a phone, tablet or PC** signed in to the web page: on the page's Radio tab choose *This device's microphone*.
- **None**: TALK then only turns the music down.

It works whether or not the radio, a cue or the soundboard is playing.

Processing

The announcement microphone is processed all the time, so it is already settled when you press TALK. The chain is:

Stage	Purpose

Low-cut (default 100 Hz)	Removes rumble and the low notes that start feedback.
Feedback suppression	Finds a tone that keeps ringing and cuts a narrow notch there (up to 6, learned in the room). Voices are not mistaken for feedback. **Forget learned feedback notches** clears them.
Noise gate	Turns the room down between words. By default it follows the room's noise by itself; untick *Gate follows the room noise* to use the manual dB value.
Auto gain	Brings every voice to a steady, loud level: target level (default -20 dB), up to a **maximum boost** (default 36 dB). It only moves while someone is speaking, so the room noise is not pumped up.
Compressor	Evens out the voice.
Limiter	Nothing louder than the **ceiling** (default -1 dB).

Every stage can be switched off, and **Process the announcement mic** switches the whole chain off. Press **Apply microphone settings** to save and use the changes.

Checking it

The Radio window and the phone page show a live level bar and a sentence about the microphone, for example the levels and gain, or *Input 3 is not open on the audio device*. If TALK does not work, read that sentence first.

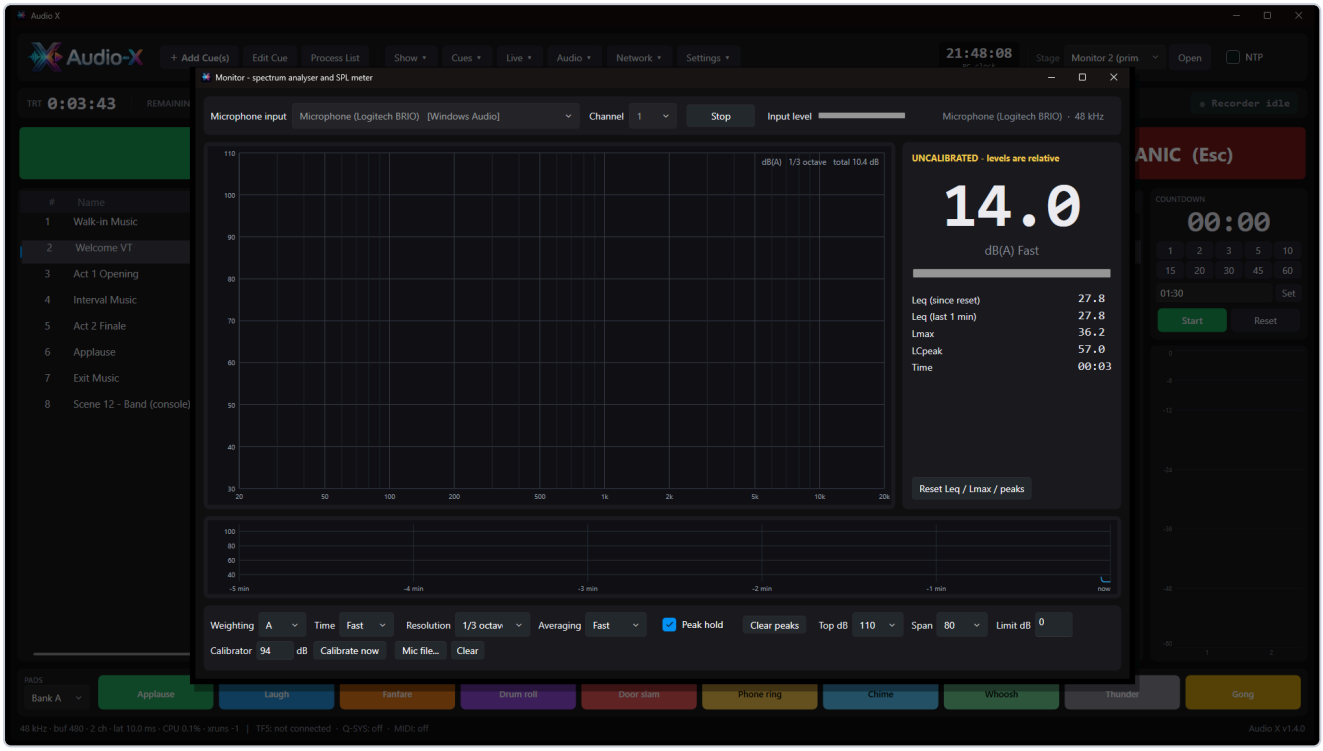
A phone or tablet microphone needs the secure page (https) because browsers only allow microphones there. See Remote Control and the Phone App.

RTA / sound level microphone

The microphone for the analyser and sound level meter. Choose a separate capture device (for example a measurement microphone) or a channel of the main device, and its **input channel**. The analyser is restarted when you press Apply if the Monitor is running.

Monitor: SPL and RTA

Live > Monitor is a real-time spectrum analyser (RTA) and a sound level (SPL) meter.



Microphone

The microphone and channel are chosen in **Audio > Channels and microphones** (the boxes at the top of the Monitor window show the choice). Press **Start** to begin. The **input level** bar and the CLIP flag show the raw microphone level: keep it out of the red.

SPL meter

- **Weighting** A, C or Z and **time weighting** Fast (125 ms) or Slow (1 s).
- Readouts: current level, **Leq** since reset, **Leq over the last minute**, **Lmax**, **LCpeak**, elapsed time, and a 5-minute history graph. **Reset Leq / Lmax / peaks** starts again.
- **Limit dB**: when the reading goes above it, the number and bar flash red, the window shows OVER LIMIT, and the main window's status bar says OVER LIMIT too.

Calibration

Put a calibrator (94 dB at 1 kHz is common) on the microphone, enter its level and press **Calibrate now**. Until then the window says *UNCALIBRATED - levels are relative*. The offset is saved. **Mic file** loads a frequency response file (for example the UMIK-1 calibration text) to correct the spectrum.

Spectrum analyser

1/3, 1/6, 1/12, 1/24 or 1/48 octave bands from 20 Hz to 20 kHz, A / C / Z weighting, averaging (Off, Fast, Slow, Long), **Peak hold**, and adjustable top level and span. Move the cursor over the graph to read a value.

Also on the web

The analyser is shown (view only) on the RTA tab of the phone page and as a tile on the confidence monitor.

Limits

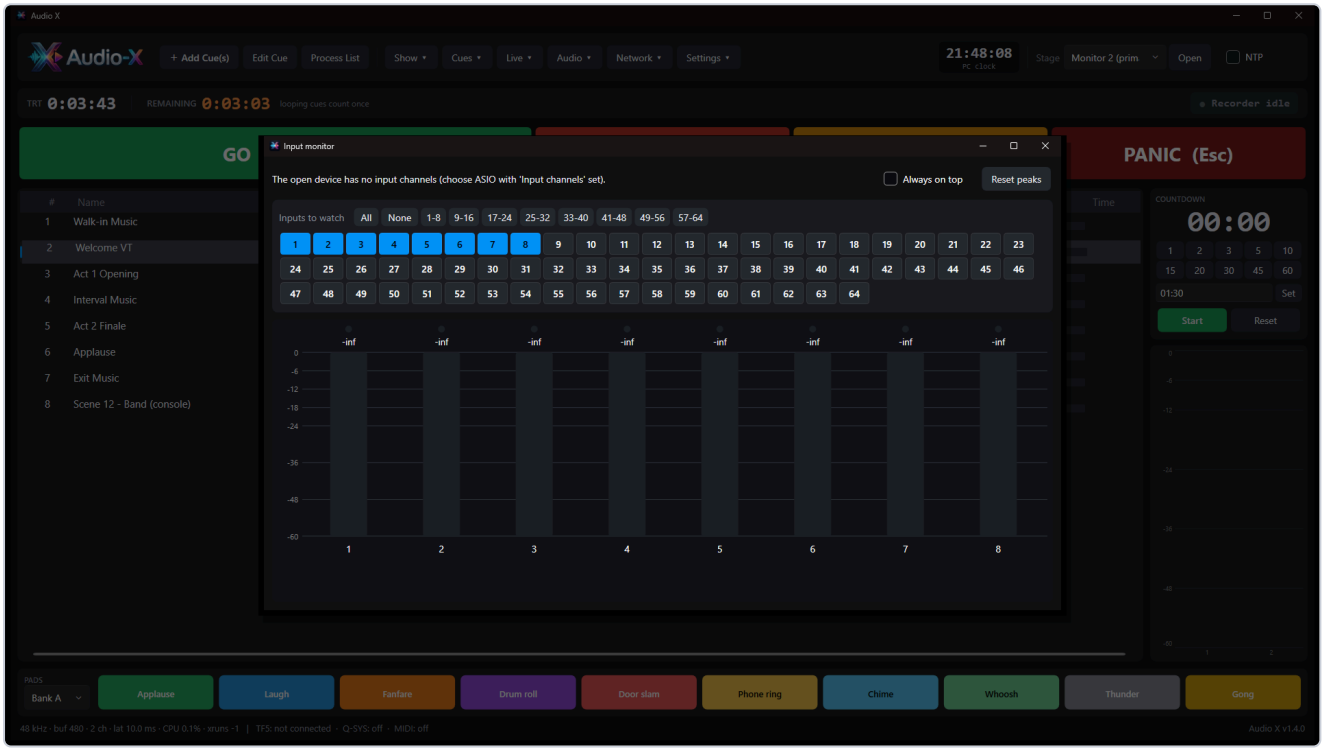
This is **not a certified sound level meter**. Accuracy depends on the microphone and calibration. Use a measurement microphone for real work (a laptop or webcam microphone has automatic gain) and a certified meter where a legal limit applies.

Input Monitor and Recorder

Both need an audio device with **input channels** open (for example *ASIO* > *Dante Virtual Soundcard* or the *Yamaha Steinberg USB* driver, with *Input channels* set in **Settings > Audio device**).

Input monitor

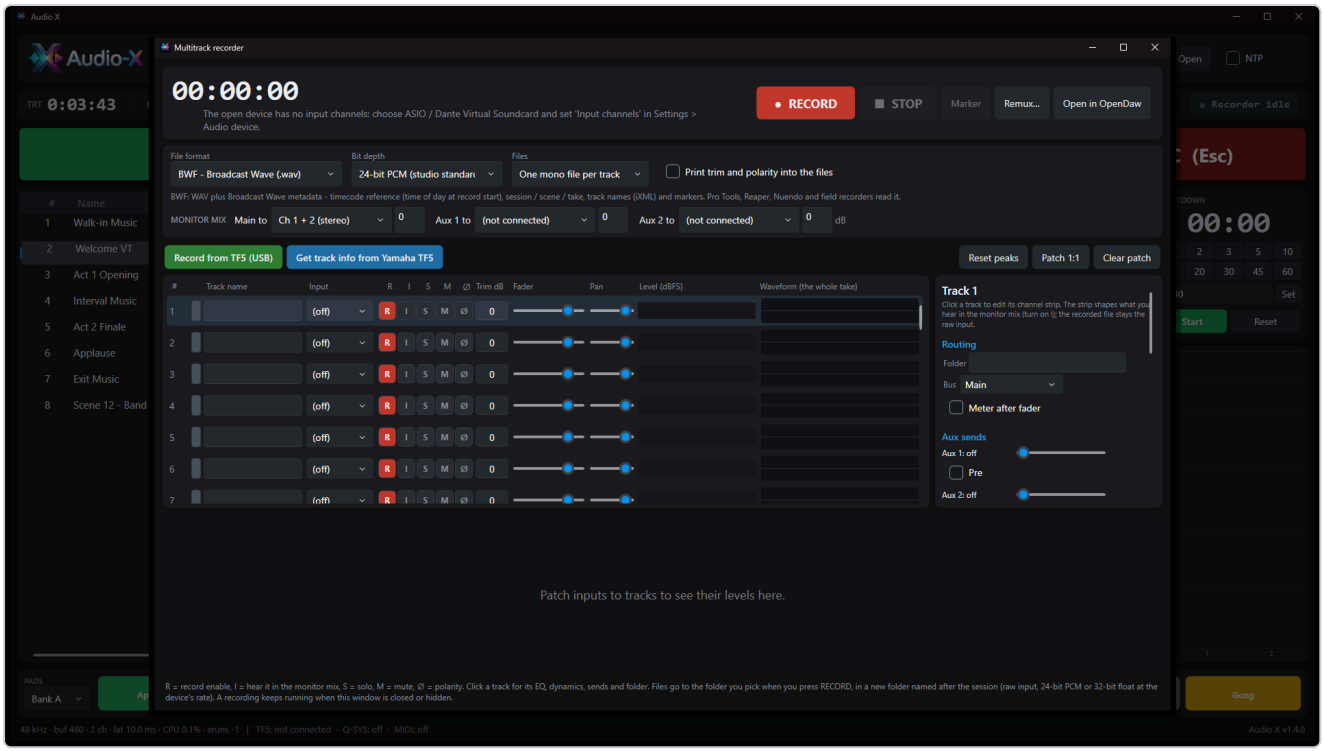
Live > Input monitor shows level bars (-60 to 0 dBFS, peak hold, clip light) for the inputs you choose.



- Tick the inputs to show, use the 8-channel range buttons, **All** or **None**.
- **Reset peaks** (or click the bars) clears the peak lines and clip lights. **Always on top** keeps the window above others.
- Channel names come from the recorder's patch (including names read from a Yamaha TF console). Click the number under a bar to change its colour.

Multitrack recorder

Live > Multitrack recorder records up to **48 tracks**.



Patching

Each track is patched to an input channel of the main device with the track's *Input* drop-down. **Patch 1:1** patches track n to input n; **Clear patch** empties the table. Meters for every patched track are live as soon as the window opens.

For each track you have **R** (record enable), **I** (input monitoring), **S** (solo), **M** (mute), **Ø** (polarity), trim, fader, pan, a colour and a name. Click a track to open its strip: bus, aux sends, high-pass, 4-band EQ, gate, compressor and limiter. These shape what you **hear** in the monitor mix only: **the recorded files are always the raw input** (an option can print trim and polarity into them).

Names and colours from a Yamaha TF console

Get track info from Yamaha TF5 reads the channel names and colours from the console (its address is set in **Settings > Show control**). Names become the file names. If you press **Record from TF5 (USB)** and the address is known, names are read automatically.

Recording

1. Press **RECORD**. Choose the folder, session name, scene, take and a note.
2. Audio X creates a folder `SessionName_yyyy-MM-dd_HHmmss`, writes all the files there and adds `session.json` when you stop.
3. **Marker** drops a cue point into every file. The main window shows a blinking **REC hh:mm:ss** badge; click it to return to the recorder.

A recording keeps going if you close the window, PANIC does not touch it, and Audio X will not close while a recording is running.

Formats

Choice	Notes
WAV	Most compatible. A file that passes 4 GB converts itself to RF64.
BWF	Broadcast Wave with time reference, iXML (project, scene, take, note, track names) and markers.
RF64 / BW64	64-bit WAV for very long recordings.
24-bit PCM or 32-bit float	32-bit float is a bit-exact copy of the input and does not clip above 0 dBFS.
Mono files or one polyphonic file	Files are named `NN_TrackName.wav`.

The sample rate is the open device's rate. File headers are refreshed every two seconds, so a crash or power cut still leaves playable files. If the disk ever stalls for longer than the buffer (about 5 seconds at 48 kHz), whole blocks are dropped for all tracks together so they stay in sync, and the window shows DROPOUTS.

Remux converts a finished session to another container, bit depth or file layout without touching the originals. **Open in OpenDaw** opens the session for editing and mixing (see Audio Tools).

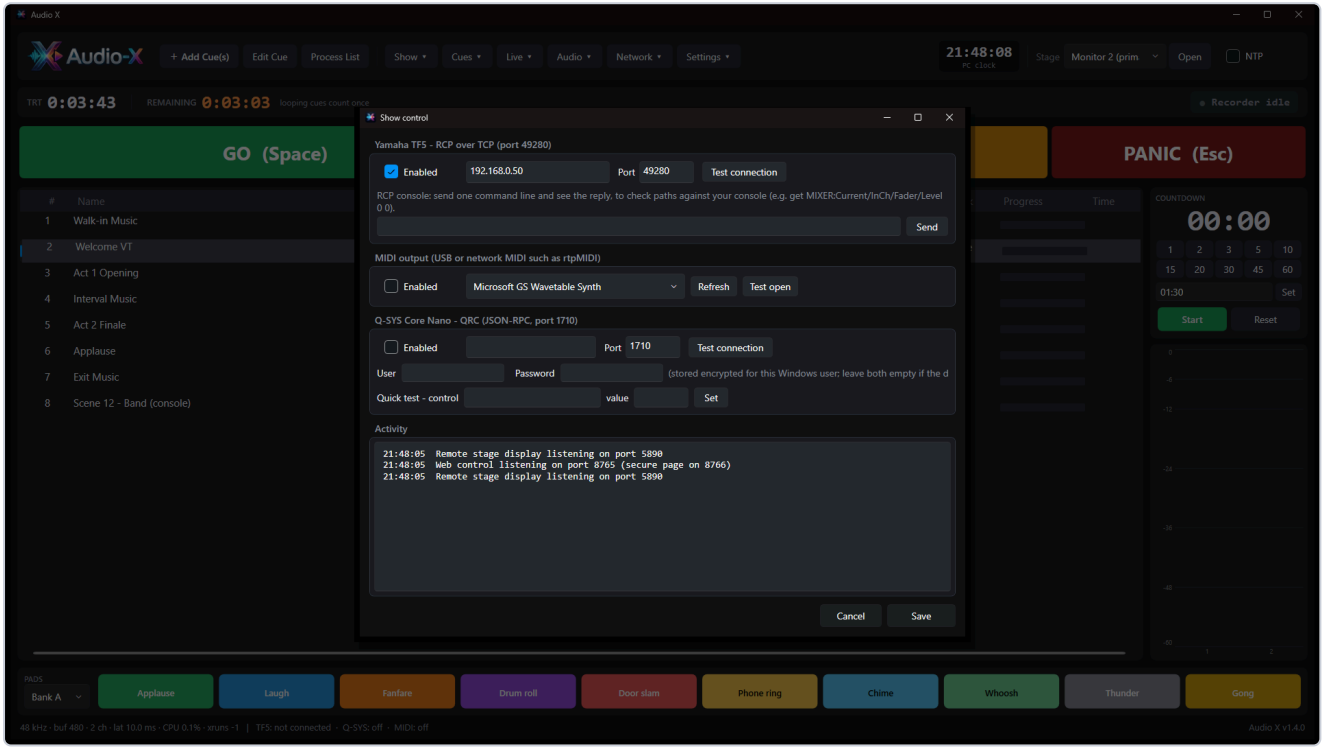
Recording straight from a Yamaha TF over USB

Install the **Yamaha Steinberg USB Driver** once, then connect the console by USB. Press **Record from TF5 (USB)**: Audio X opens the driver at 48 kHz with all 34 inputs and patches USB n to track n. Only one program can use an ASIO driver at a time, so close any other program that has the console open.

Show Control: Yamaha, MIDI and Q-SYS

A cue can do things to other equipment when it fires. The equipment is set up once in **Settings > Show control**, then each cue gets **actions**.

Setting up the equipment



- **Yamaha TF** (RCP over TCP, port 49280): tick *Enabled*, enter the console's IP address and press **Test connection**. The RCP box sends one command line and shows the reply, to check paths against your console (for example `get MIXER:Current/InCh/Fader/Level 0 0`).
- **MIDI output**: tick *Enabled*, choose a Windows MIDI output (USB or network MIDI such as rtpMIDI), **Refresh** the list and **Test open**.
- **Q-SYS Core** (QRC, port 1710): address, optional user and password (stored encrypted for the current Windows user) and a quick test of one named control.
- The **Activity** log shows what was sent and any problem. Press **Save**.

The bottom status bar shows the state of all three. A device that is offline is skipped for five seconds after a failure so a dead console cannot delay every cue.

Adding actions to a cue

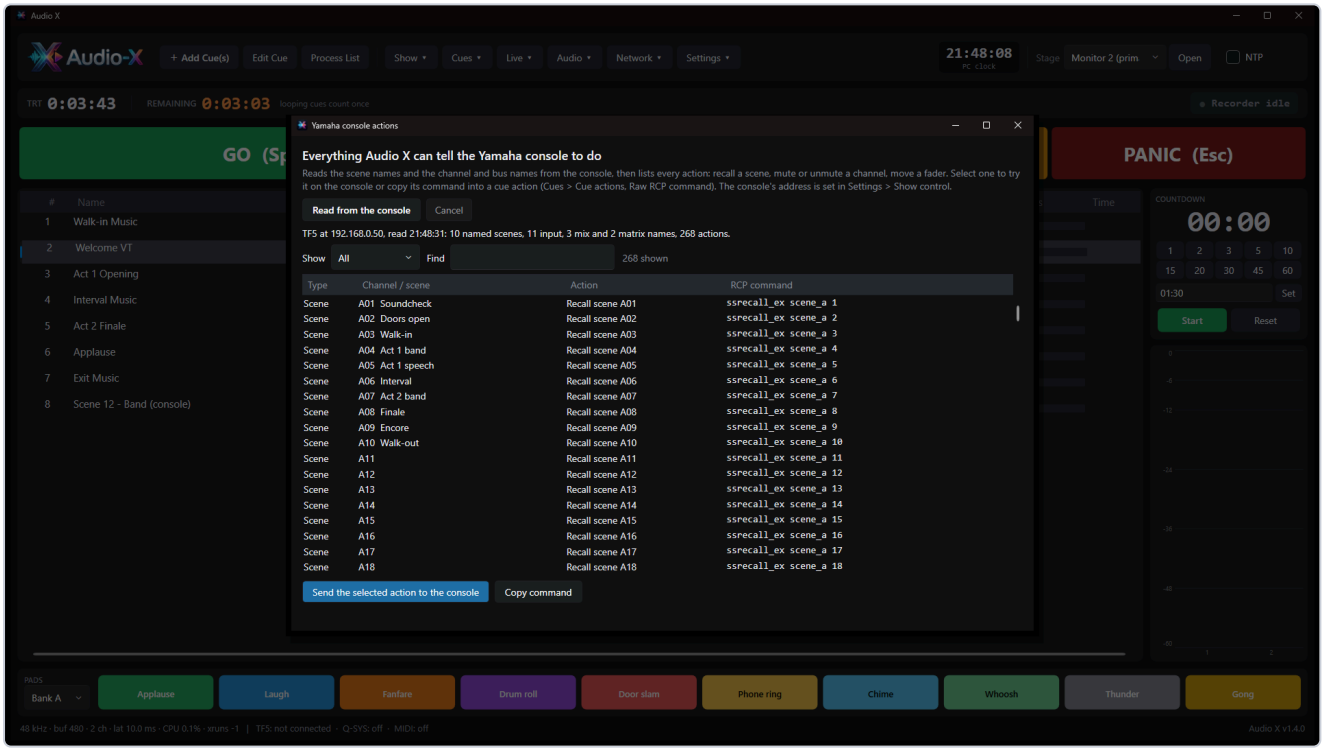
Select a cue and open **Cues > Cue actions**, or add a cue with no audio using **Cues > Add action cue**. Every action has a delay in milliseconds; all actions of a cue start when the cue fires, each on its own timeline.

Action	What it does

Yamaha scene	Recalls scene A or B, number 1-100.
Yamaha fader	Sets Input 1-40, Mix 1-20, Matrix 1-4 or the Stereo master to a level from -138 to +10 dB, or -inf.
Yamaha mute / unmute	Mutes or unmutes a channel or bus.
Yamaha raw	Sends any RCP line exactly as typed.
MIDI	Program change (with optional bank), control change, note (with automatic note-off) or SysEx.
Q-SYS control	Sets a named control (number, true/false or text) with an optional ramp.
Q-SYS snapshot	Loads a snapshot bank and number with an optional ramp.

Yamaha console actions (the list)

Live > Yamaha console actions shows everything Audio X can tell the console to do.



1. Press **Read from the console**. Audio X reads the product name, the scene names (banks A and B) and the input, mix and matrix channel names.
2. The table lists every action with its **exact RCP command**: recall each scene, and mute, unmute, set to 0 dB or set to -inf for every channel, bus and the master.
3. **Show** filters by type (Scene, Input, Mix, Matrix, Stereo); **Find** searches names, scene numbers and commands.
4. Select a row and press **Send the selected action to the console** (it asks first, because it changes the live console), or **Copy command** and paste it into a cue action of type *Yamaha raw*.

If the console does not give scene names, the recall actions are still listed, without names, and the window says so.

Check it on your own console

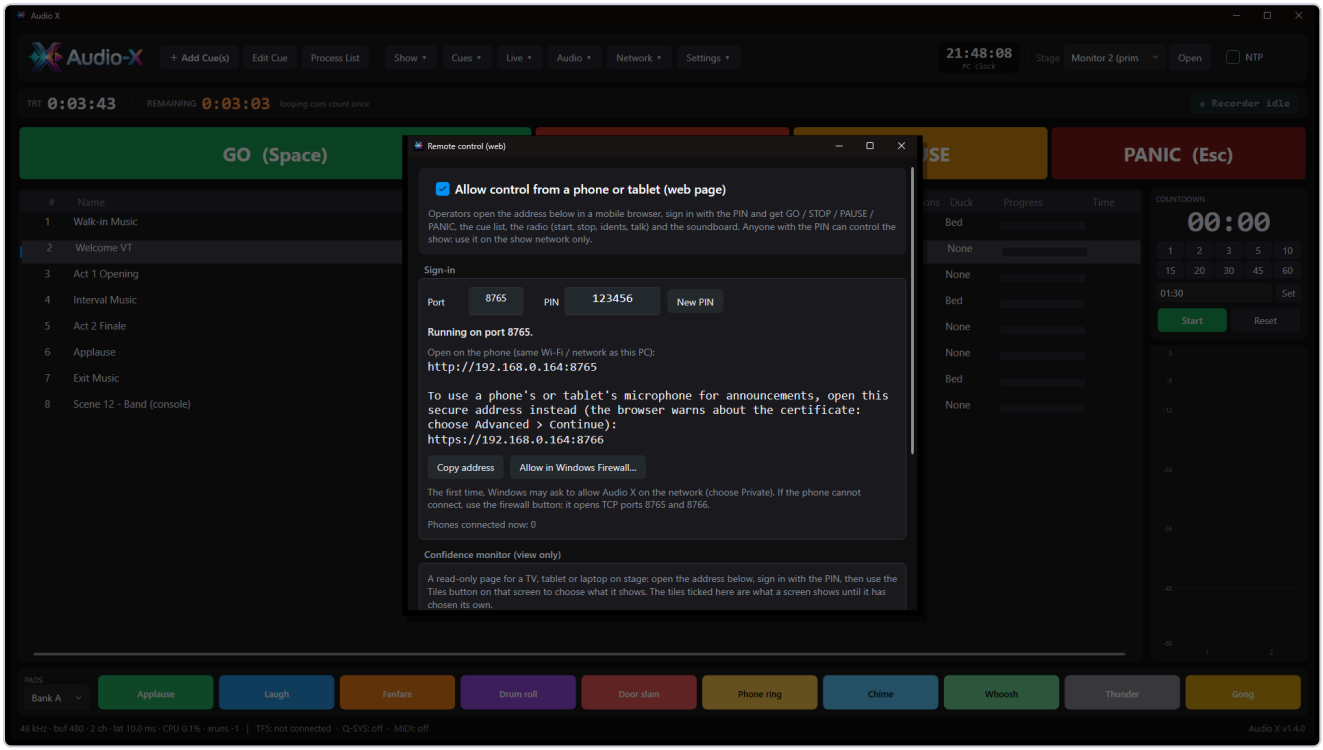
The command paths follow the published Yamaha RCP conventions and were tested against a simulated console. Before a show, use the RCP box in Show control (or this list's **Send** button) to confirm the scene recall and the master fader path on your own TF, and use a *Yamaha raw* action for anything that differs. MIDI scene recall depends on the console's Program Change settings.

Remote Control and the Phone App

Control Audio X from a phone, tablet or another PC on the same network.

Switch it on

Network > **Remote control**: tick *Allow control from a phone or tablet*.

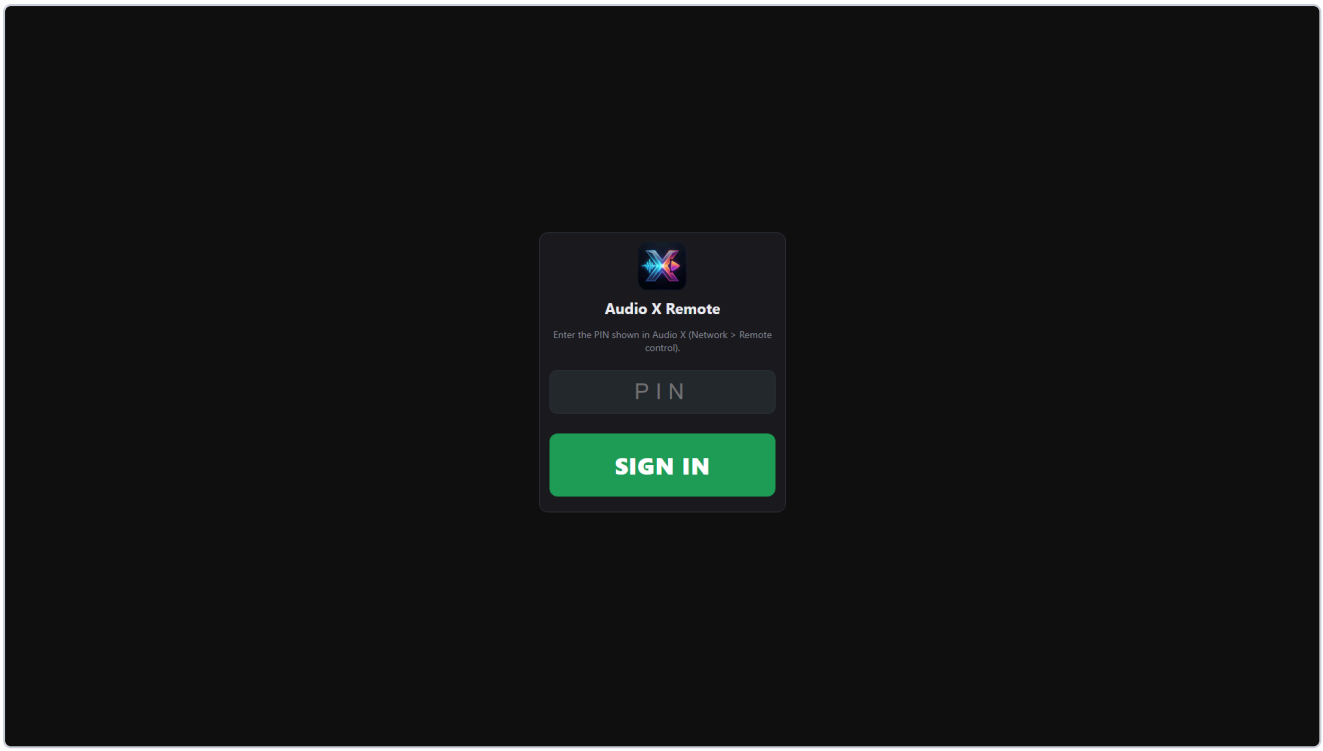


- **Port** (default 8765) and **PIN**. **New PIN** makes a new one and signs everybody out.
- The window shows the address to open, for example `http://192.168.0.164:8765`, and the **secure address** on the next port (`https://...:8766`) that you need to use a phone's microphone.
- **Allow in Windows Firewall** adds a rule for both ports (it asks for administrator permission).
- *Phones connected now* shows how many are signed in.
- The **Confidence monitor** section chooses the default tiles of the view-only confidence page.

Anyone with the PIN can control the show: use it on the show network only. Five wrong PIN attempts lock that address out for a minute. The plain page is not encrypted.

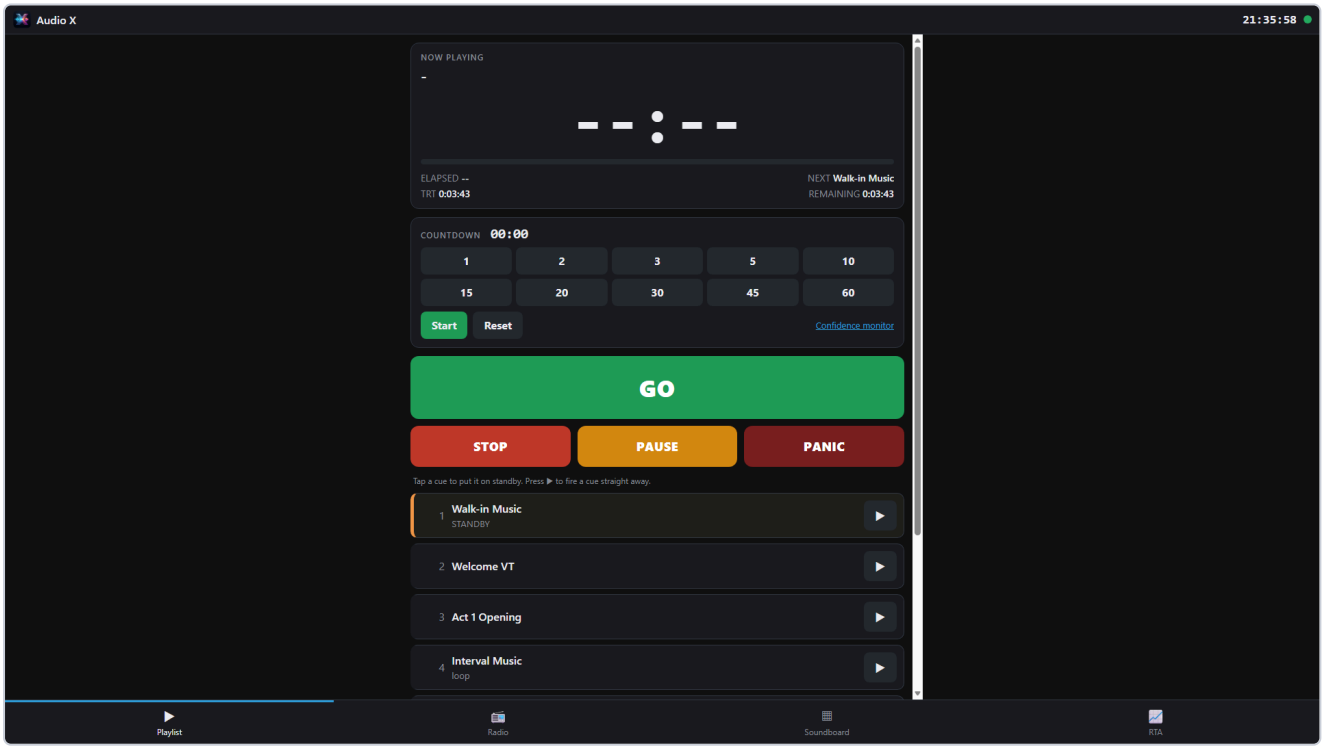
The web page

Open the address in a browser, enter the PIN. "Add to Home screen" gives a full-screen app icon. The page keeps the screen awake and reconnects by itself.



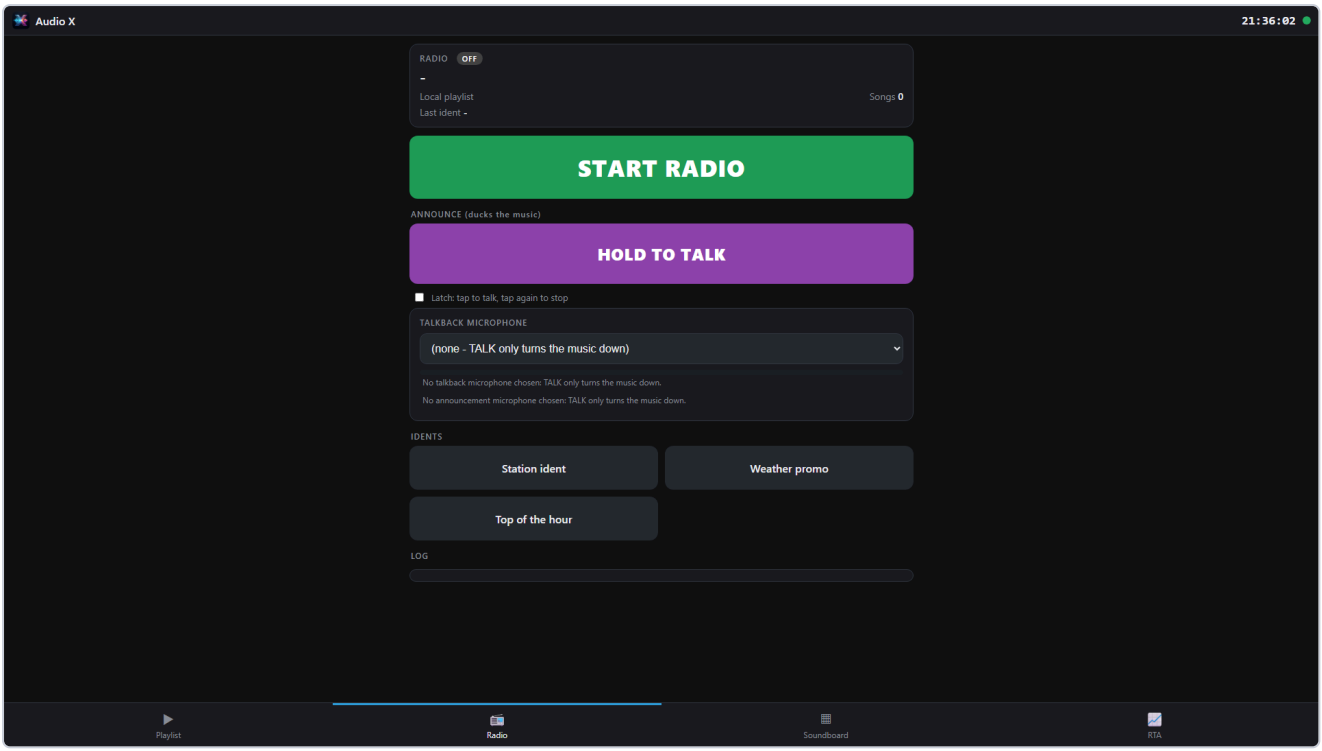
Playlist tab

GO, STOP, PAUSE and PANIC, the now-playing display with elapsed, remaining and next, the countdown timer, and the cue list: tap a cue to put it on standby, tap its play button to fire it.



Radio tab

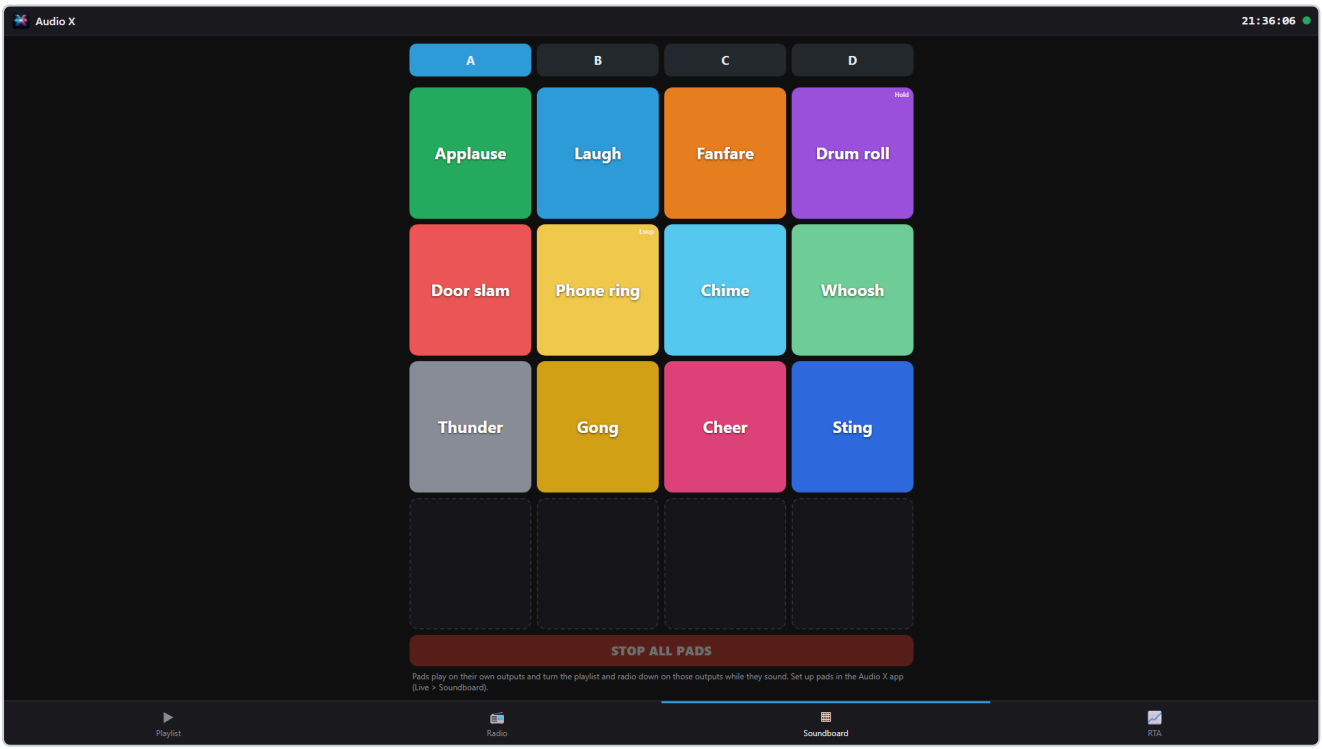
Start and stop the radio, see what is playing, **HOLD TO TALK** (or tick *Latch* to tap to talk and tap to stop), the **Talkback microphone** chooser with its level bar and status line, and a button for every ident.



To use this device's own microphone for announcements, open the **secure** address (`https://...:8766`), accept the browser's certificate warning once, sign in, and choose *This device's microphone* in the Talkback microphone list. The page then streams the microphone to Audio X while TALK is on.

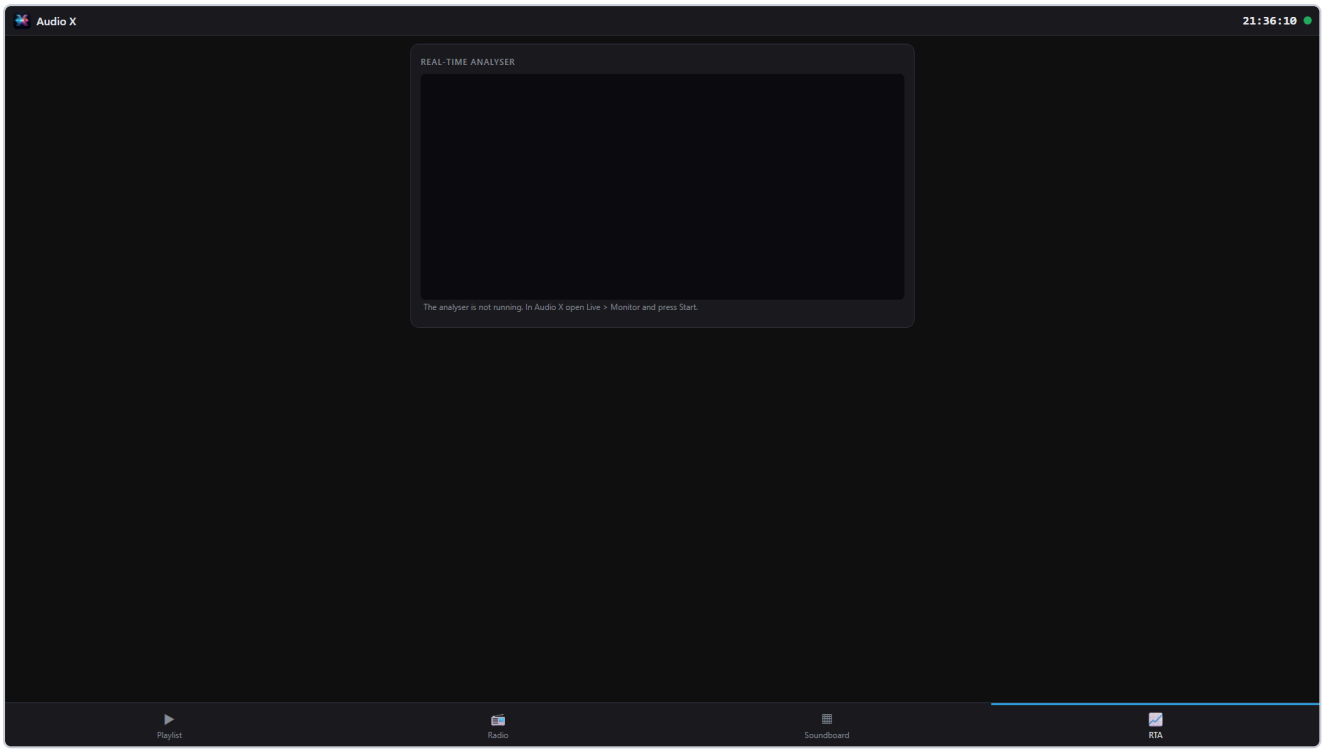
Soundboard tab

All 16 pads of the chosen bank, banks A to D, **Stop all pads**, and a note of how far the playlist and radio are being ducked. Hold pads play only while pressed.



RTA tab

A view-only copy of the analyser. It shows a message if the analyser is not running (start it in **Live > Monitor**).



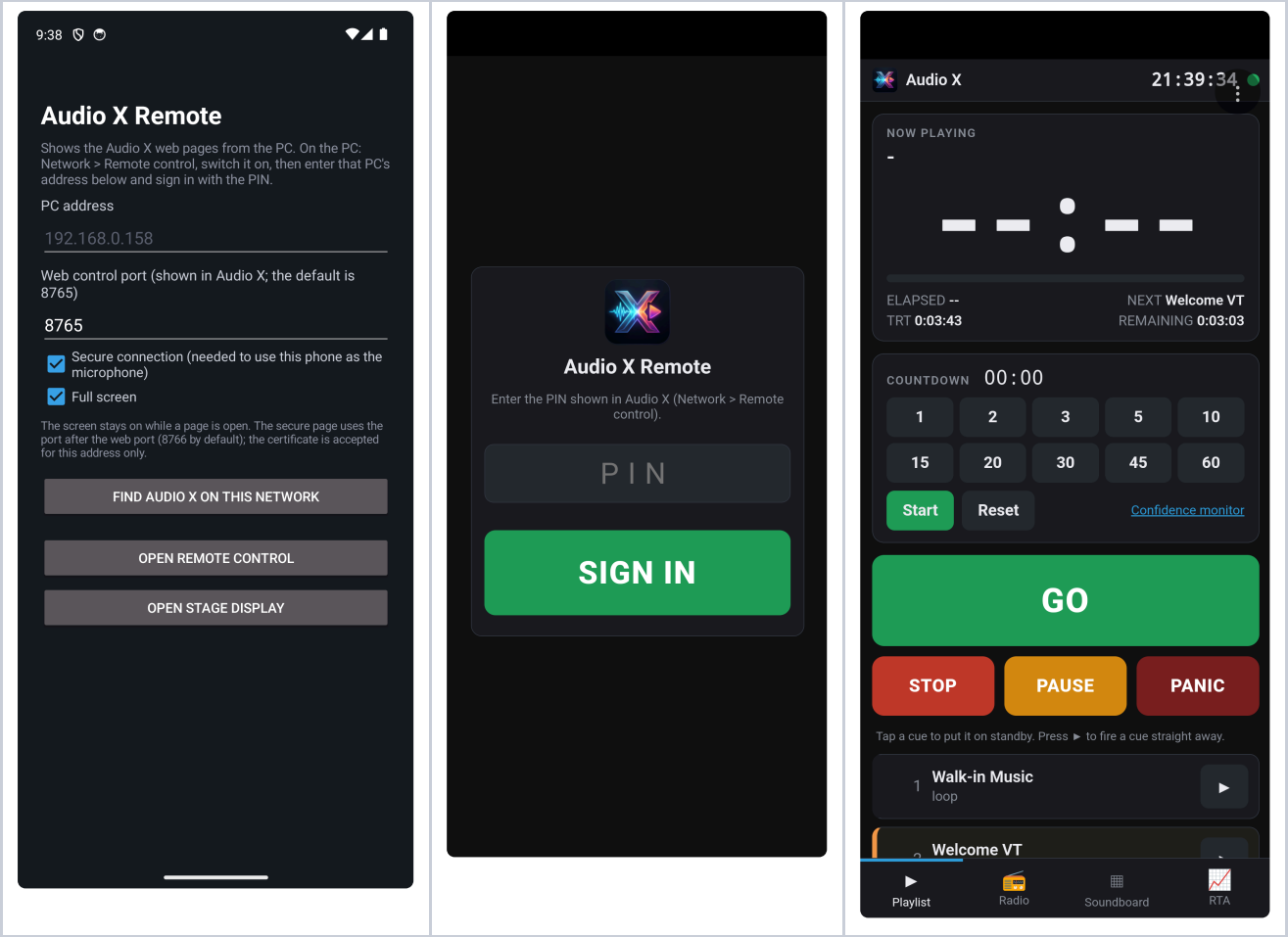
The confidence monitor

`http://<pc>:8765/confidence` (or `/stage`) is a view-only page for a TV or laptop on stage. Use its **Tiles** button to choose what it shows. See Stage Display.

The Audio X Remote phone app (Android)

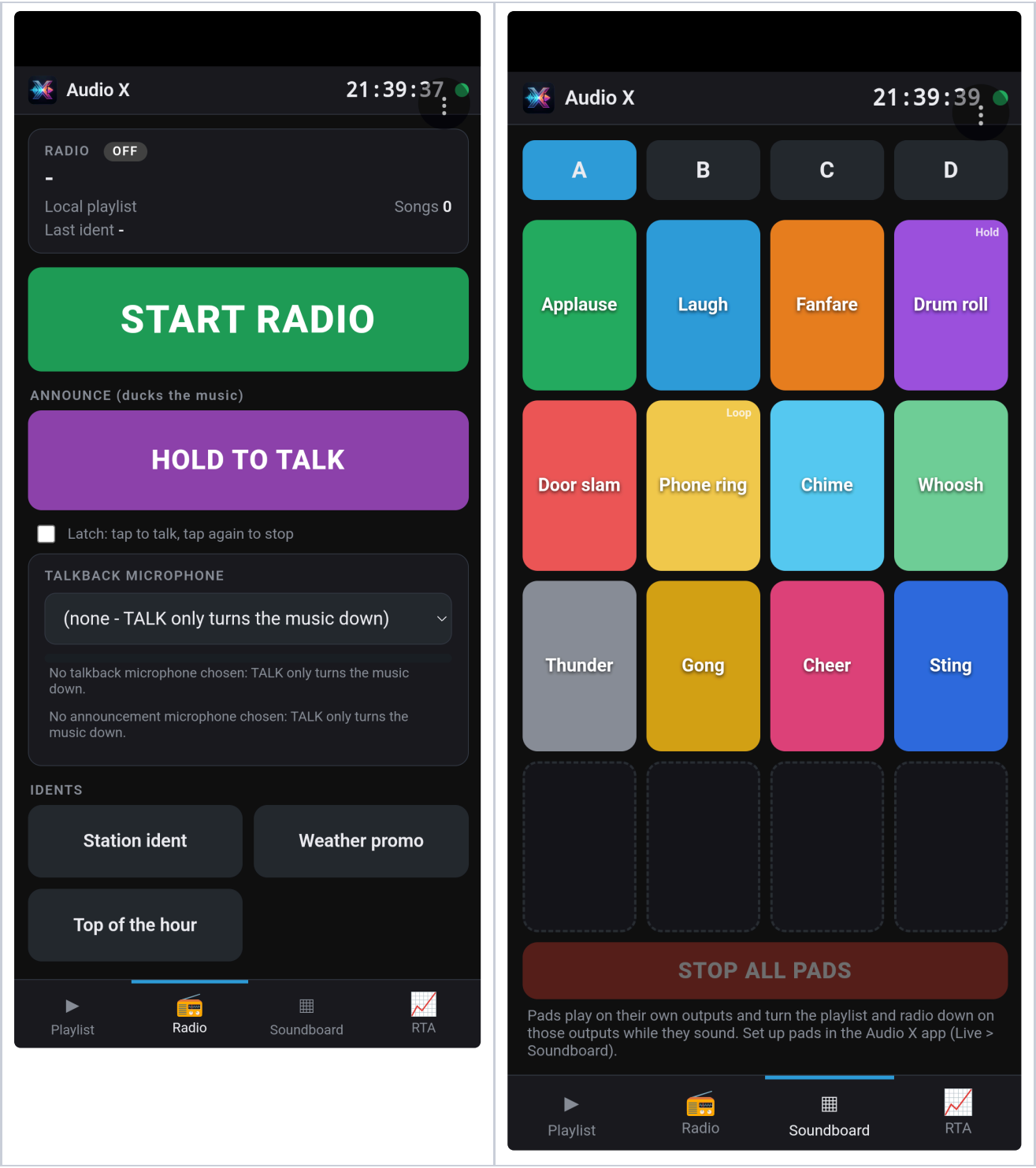
`AudioXRemote-1.0.1.apk` is a small Android app that shows these same pages full screen.

Setup	Sign in	Playlist



1. Copy the APK to the phone and open it (allow *install unknown apps* the first time).
2. Enter the PC's address and the web port (8765). **Find Audio X on this network** looks for the PC by itself if the stage display sharing is on.
3. Leave **Secure connection** ticked to be able to use the phone as a microphone; the app accepts the PC's certificate for that one address only. If the secure page cannot be reached, the app tries the plain page and says so.
4. Press **Open remote control** (or **Open stage display**), sign in with the PIN. The address is remembered and the screen stays on.
5. The round menu button (top right, fades after a moment) switches between remote control and stage display, reloads, or changes the PC.

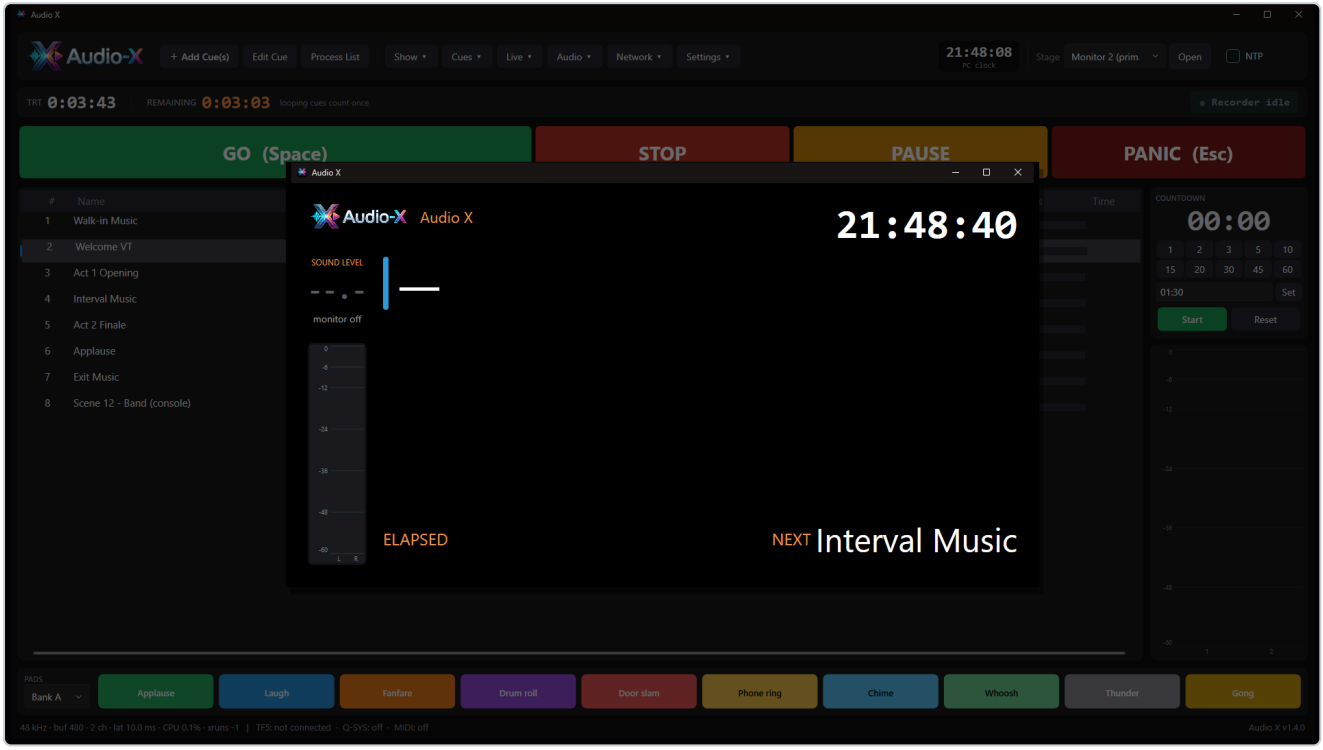
Radio	Soundboard



If the app cannot connect, it shows which of the two ports answers. Check that Network > Remote control is switched on in Audio X, that both devices are on the same Wi-Fi, and the firewall button.

Stage Display

The stage display (also called the confidence monitor) shows the show's clock, what is playing and what is next, so performers and crew can see it on any screen.



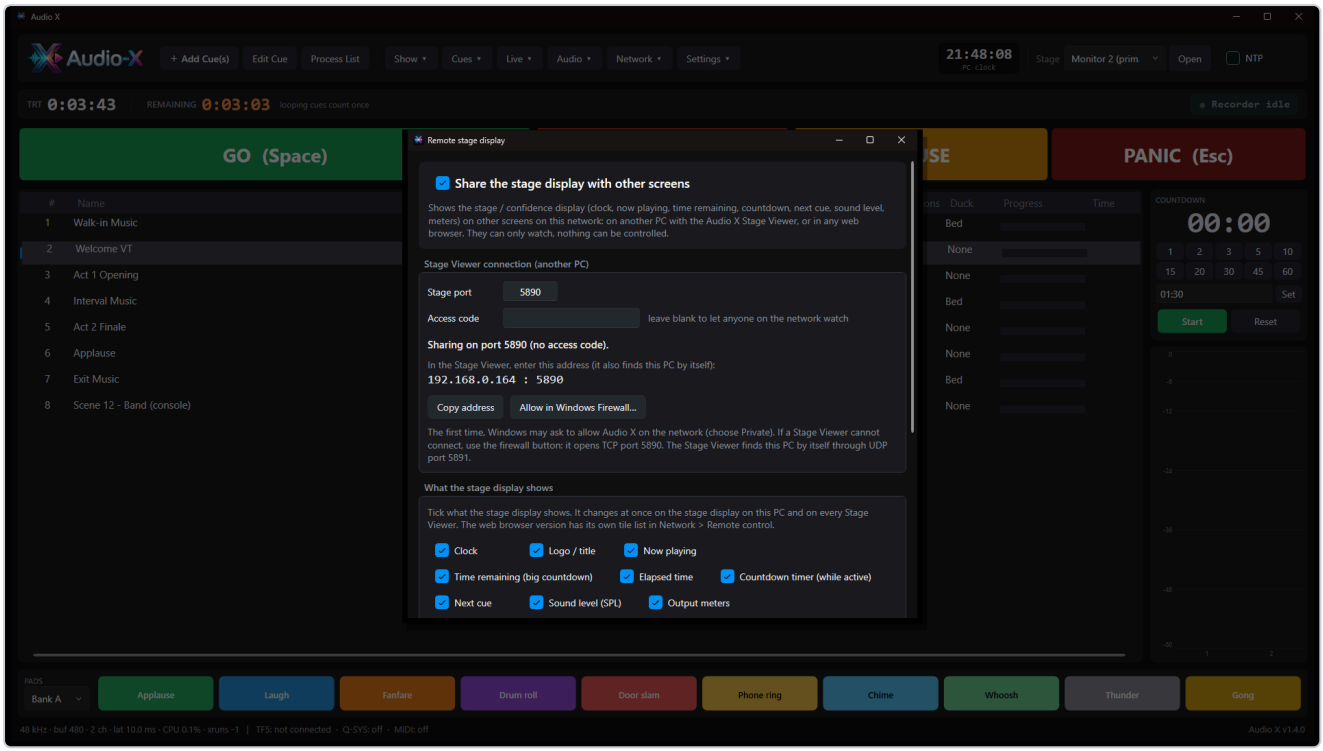
It shows the clock, the cue name, the time remaining with a red warning flash at the end, elapsed time, the next cue, the countdown timer while it runs, the sound level and stereo meters, and your logo and title (see Branding in Settings Reference).

On this PC

Choose a screen in the **Stage** box at the top of the main window and press **Open**. F11 or the window's full-screen button fills the screen.

On other PCs (Audio X Stage Viewer)

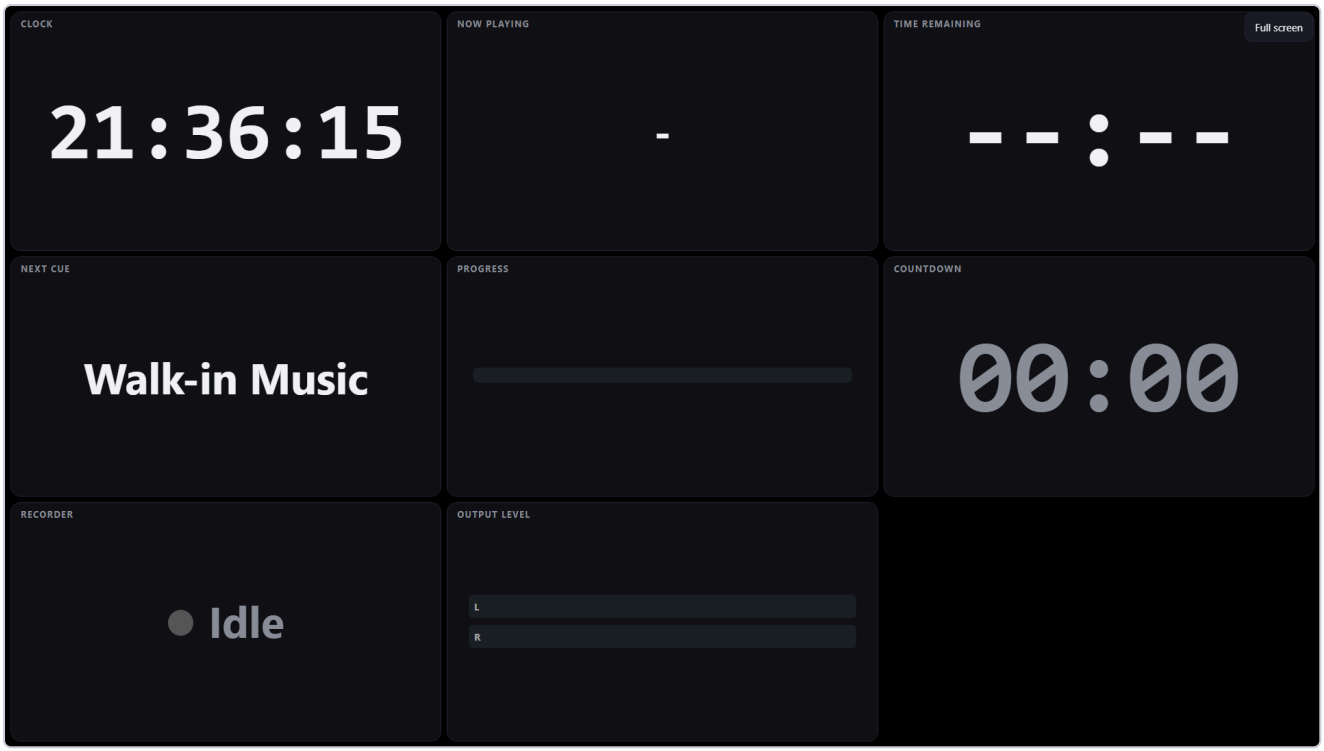
Network > Remote stage display shares the display with other PCs.



- **Stage Viewer connection (another PC)**: tick *Share the stage display with other screens*, choose the **stage port** (default 5890) and an optional **access code**. The window shows the address to enter in the Stage Viewer; the Viewer also finds this PC by itself.
- **What the stage display shows**: tick the elements (clock, logo/title, now playing, time remaining, elapsed, countdown, next cue, sound level, output meters). It changes at once on this PC and on every Stage Viewer.
- **Stage Viewers connected now** lists who is watching.

On the other PC install `AudioXStage-Setup-1.0.0.exe` (or unzip `AudioXStage-1.0.0-win-x64.zip`). It lists Audio X PCs it finds on the network, or you type the address, and it reconnects by itself after a network drop. F11 or double-click for full screen, F2 or right-click for the connection panel, Esc to leave full screen. For a kiosk screen it can be started with `AudioXStage.exe --host 192.168.1.20 --fullscreen --screen 2`. Use **Allow in Windows Firewall** if a Viewer cannot connect.

In a web browser



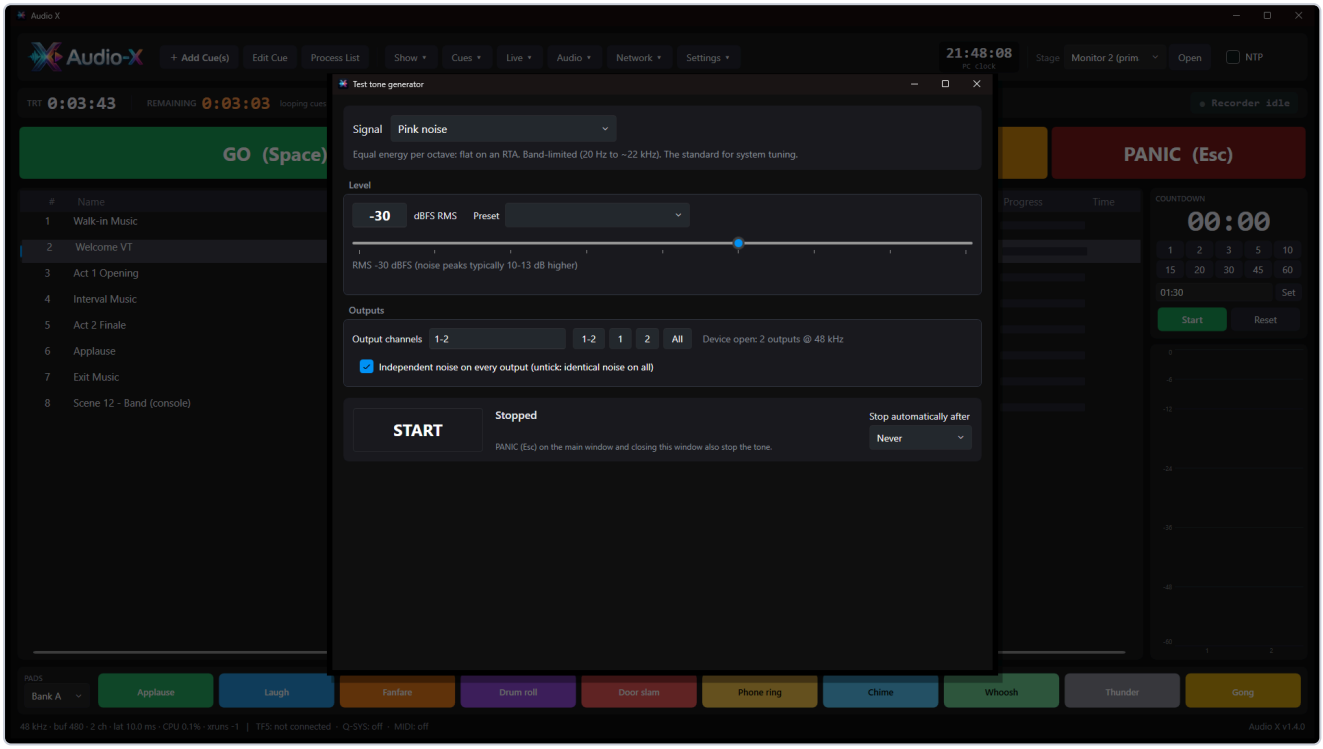
Any browser (phone, tablet, smart TV, laptop) can show it. In the **Watch in a web browser** section of the Remote stage display window, copy the address `http://<this PC>:5890`: it sends the browser on to the stage page of the web control, where you sign in with the web PIN (Network > Remote control must be on). The page's **Tiles** button chooses what it shows. The Android app has an **Open stage display** button for the same page.

The viewers can only watch, nothing can be controlled from them. The connection is not encrypted: use it on the show network.

Audio Tools

Tone generator

Audio > Tone generator makes test signals for tuning a system.

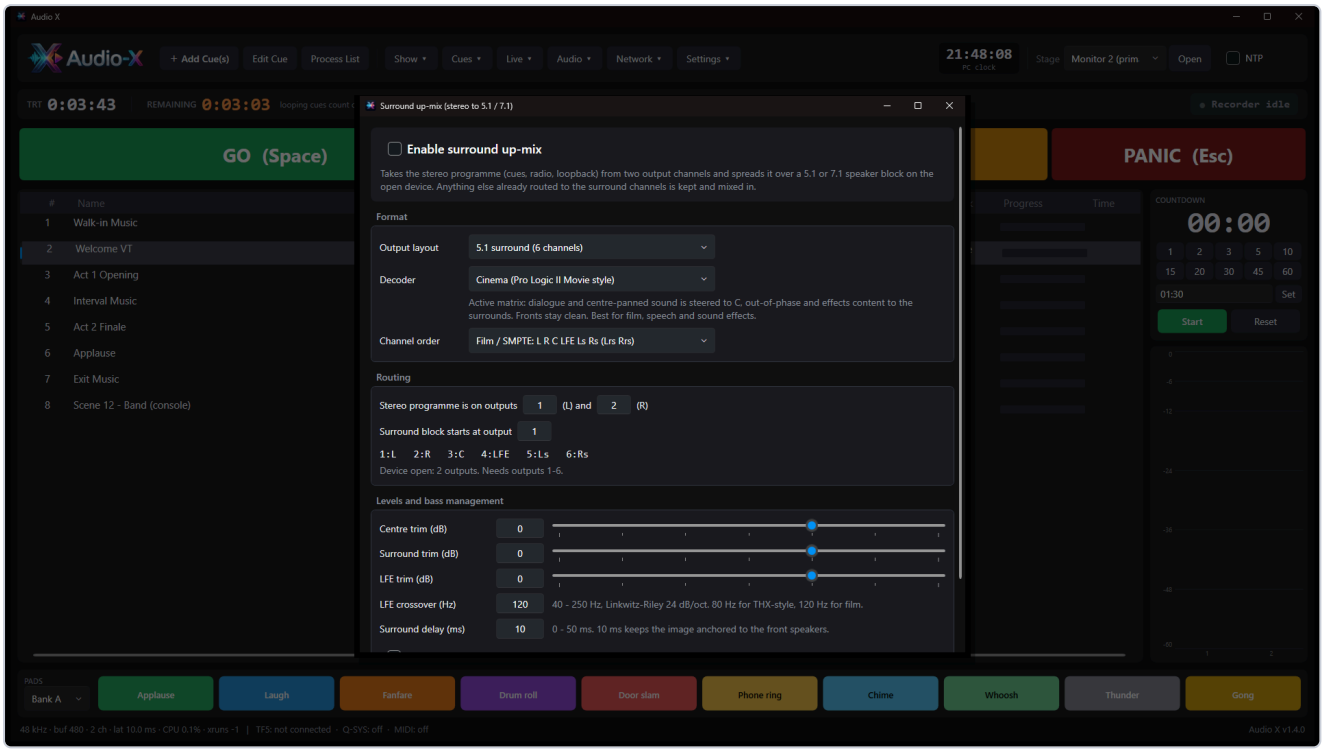


Signal	Notes
Sine, square, triangle, sawtooth	Band-limited.
Log sine sweep	Start, end and duration; repeats.
White, pink, brown, blue, violet noise	Pink is flat to within 1 dB from 160 Hz to 16 kHz.
Pink noise in a 1/3 or 1-octave band	Centred on the chosen frequency.
Pink noise bursts	On and off times with soft edges.
Click / impulse train	Clicks per second; level is peak.

- **Frequency**: type Hz, step by 1 Hz or 1/3 octave, use the slider, the octave buttons or the ISO 1/3-octave list.
- **Level** in dBFS RMS (peak for clicks), with presets for -18, -20 and -30 dBFS and a warning at -12 dBFS and above.
- **Outputs**: `1-2`, `1,2,5-8` or `all`. Noise can be independent on each output or identical.
- Changes apply live. Start and stop use a 150 ms fade. An optional timer stops it. PANIC stops the tone.

Surround up-mix (5.1 / 7.1)

Audio > Surround up-mix turns the stereo programme (cues, radio, loopback on outputs 1-2) into a 5.1 or 7.1 mix on the open device.



- **Decoder**: *Cinema* (steered centre and surround, for film and speech), *Music* (gentler steering with ambience), *Passive matrix*, or *Stereo + subwoofer only*.
- Surround delay (0-50 ms), subwoofer crossover (40-250 Hz) and optional **redirect bass**.
- Channel order: Film/SMPTE or Pro Tools.
- It needs enough outputs on the device (6 for 5.1, 8 for 7.1); the window says why if it is not active.

This is an independent up-mixer, not a Dolby or DTS decoder, so it will not sound identical to those.

OpenDaw (audio editing and mixing)

Audio X does not contain a full editor; it hands audio to **OpenDaw**, a free open-source program that you install separately.

- **Audio > OpenDaw** starts it.
- In the Cue Editor, **Edit audio in OpenDaw** opens the cue's files, and the waveform reloads when you return.
- In the Recorder, a project for the session is written when you stop, and **Open in OpenDaw** opens it.

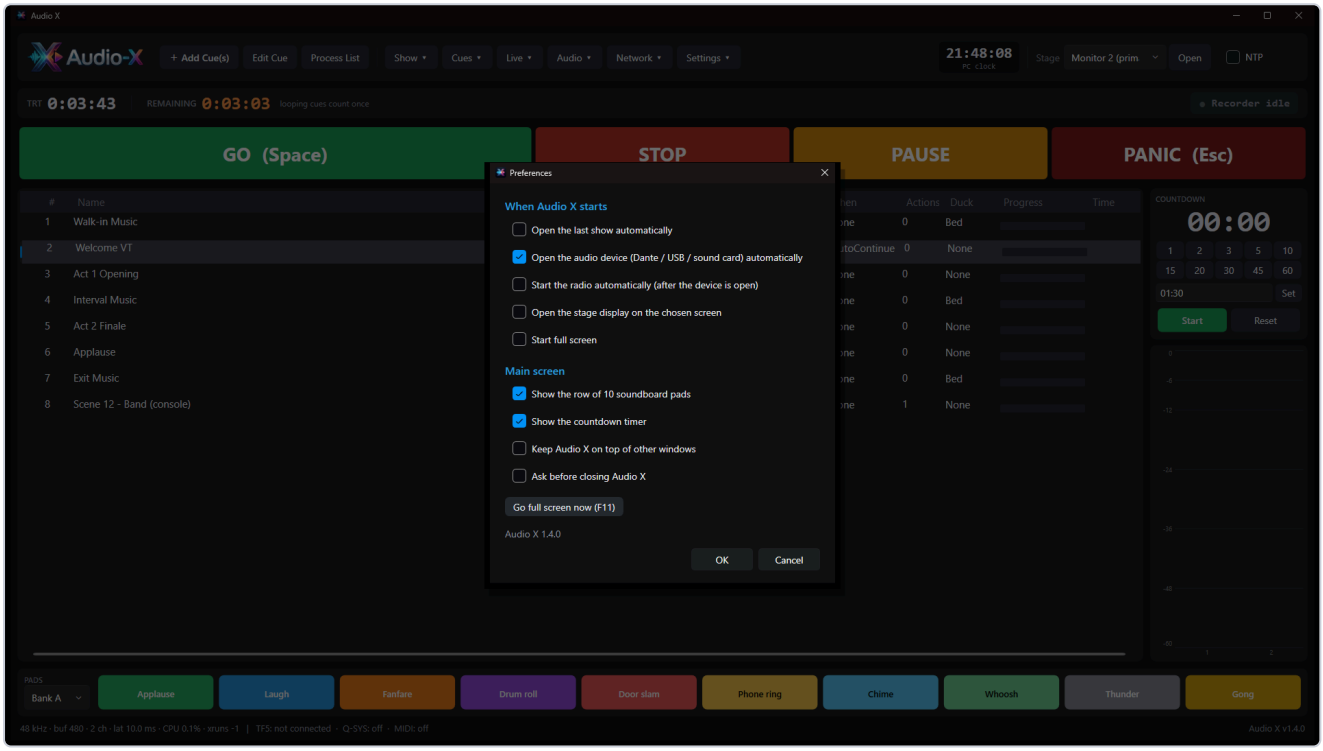
If Audio X cannot find OpenDaw, it offers *Open download page* and *Locate OpenDaw.exe*. The generated project files could not be tested in a real OpenDaw; if one opens empty, drag the WAV files from the session folder into OpenDaw. Your WAV files are never changed.

Settings Reference

Audio device (Settings > Audio device)

Opens the driver and sets rate, buffer and channels. See Getting Started.

Preferences



When Audio X starts: open the last show, open the audio device, start the radio (after the device is open), open the stage display on the chosen screen, start full screen.

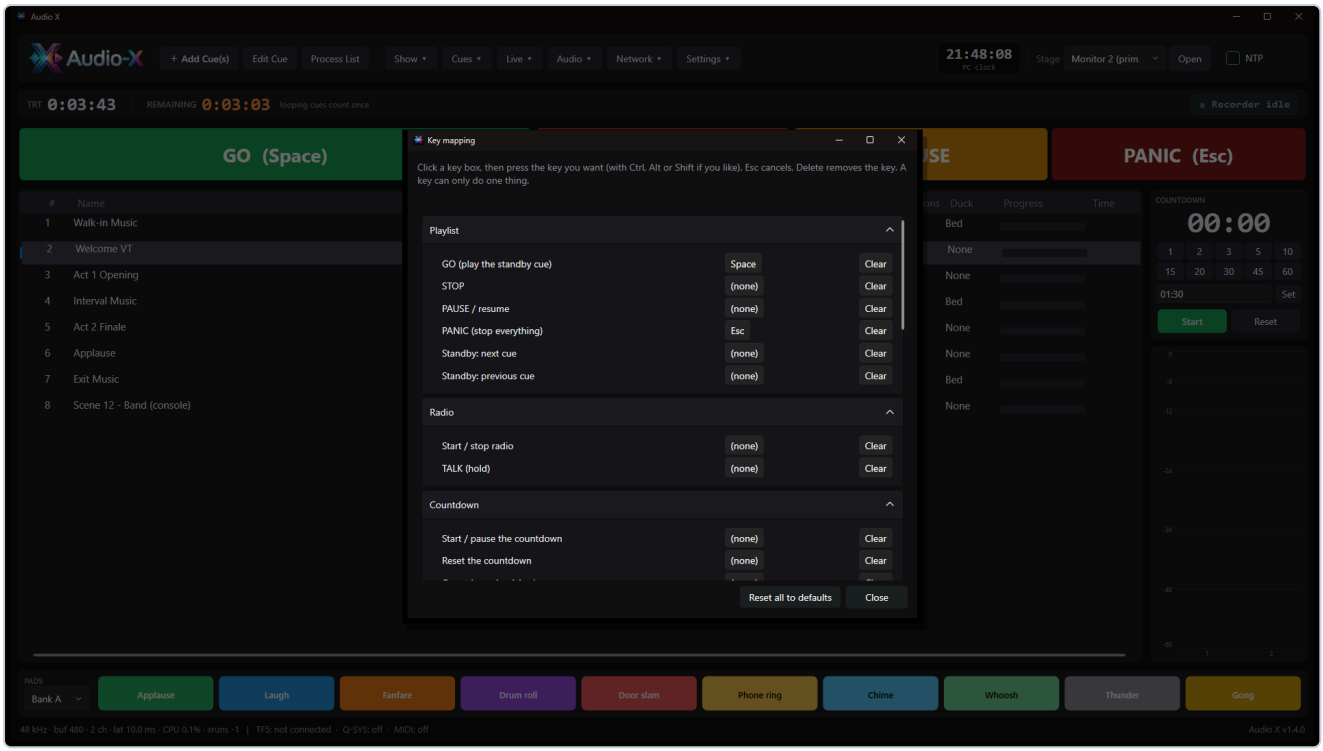
Main screen: show the row of 10 soundboard pads, show the countdown timer, keep Audio X on top of other windows, ask before closing.

Go full screen now (F11) switches full screen on and off.

Hardware acceleration

GPU and parallel-job settings. See Hardware Acceleration.

Key mapping

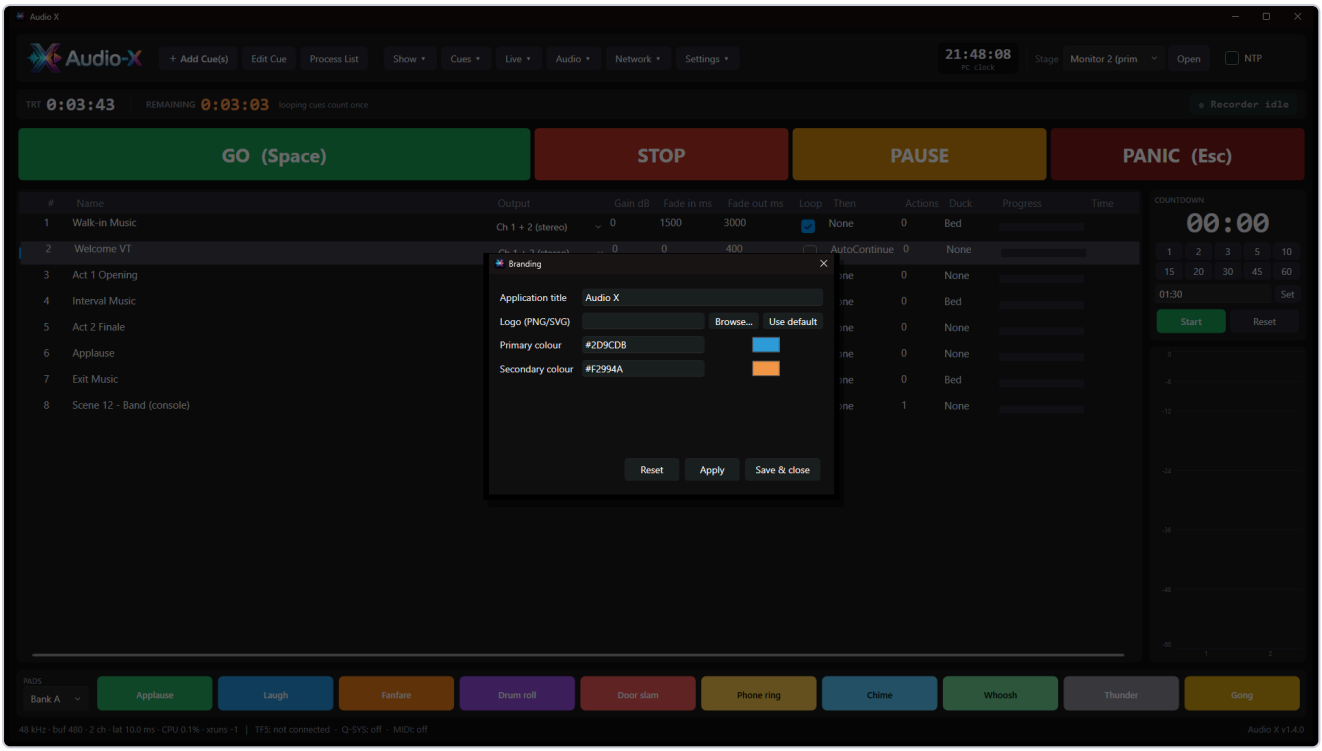


Click a key box, then press the key you want (with Ctrl, Alt or Shift if you like). Esc cancels, Delete removes the key. A key can only do one thing. You can map GO (default Space), STOP, PAUSE, PANIC (default Esc), standby next and previous cue, radio start/stop, TALK (hold), the countdown, markers, any of the ten main-row pads, any of the 64 pads, and the 16 pad keys of the soundboard window. **Reset all to defaults** restores the defaults.

Show control

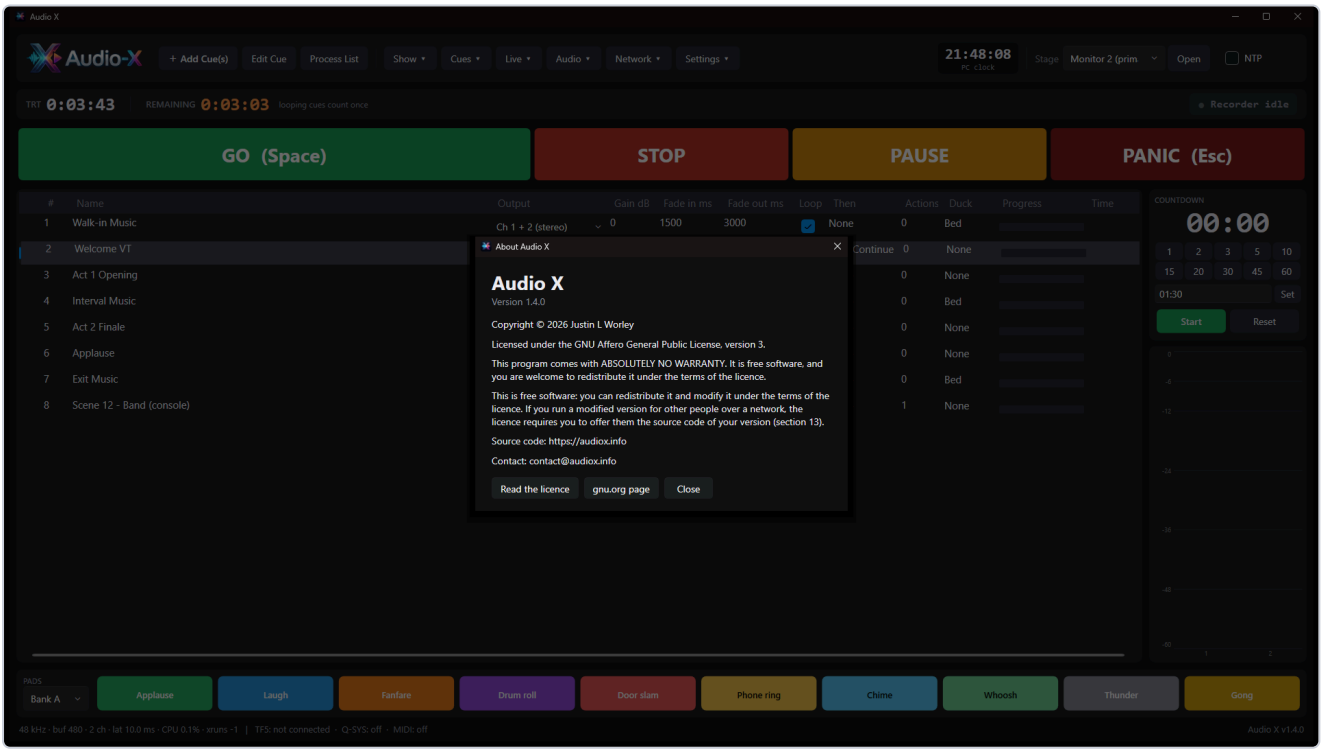
Yamaha, MIDI and Q-SYS equipment. See Show Control.

Branding (Settings > Branding)



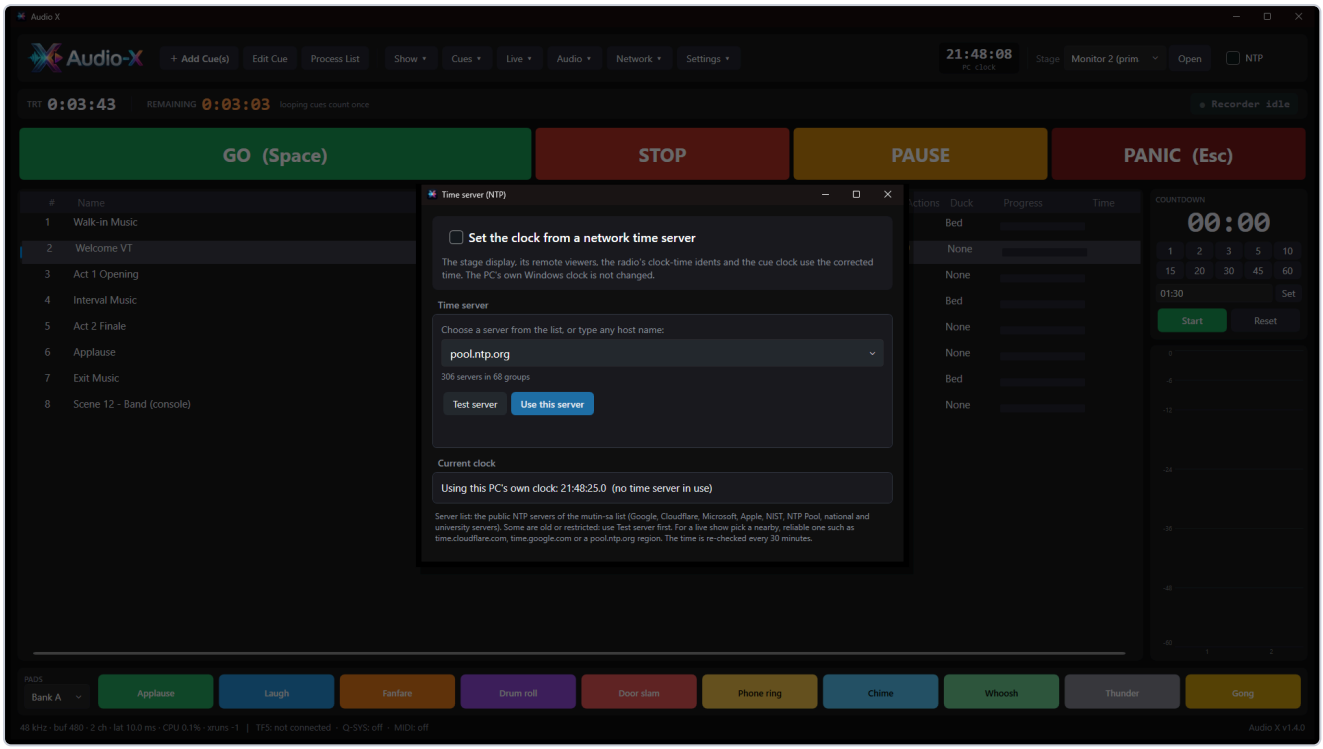
Sets the **application title**, the **logo** (PNG or SVG; *Use default* restores Audio X's) and the **primary and secondary colours**. The title, logo and colours appear in the main window and on every stage display. **Apply** previews, **Save & close** keeps, **Reset** restores the defaults.

About and licence (Settings > About and licence)



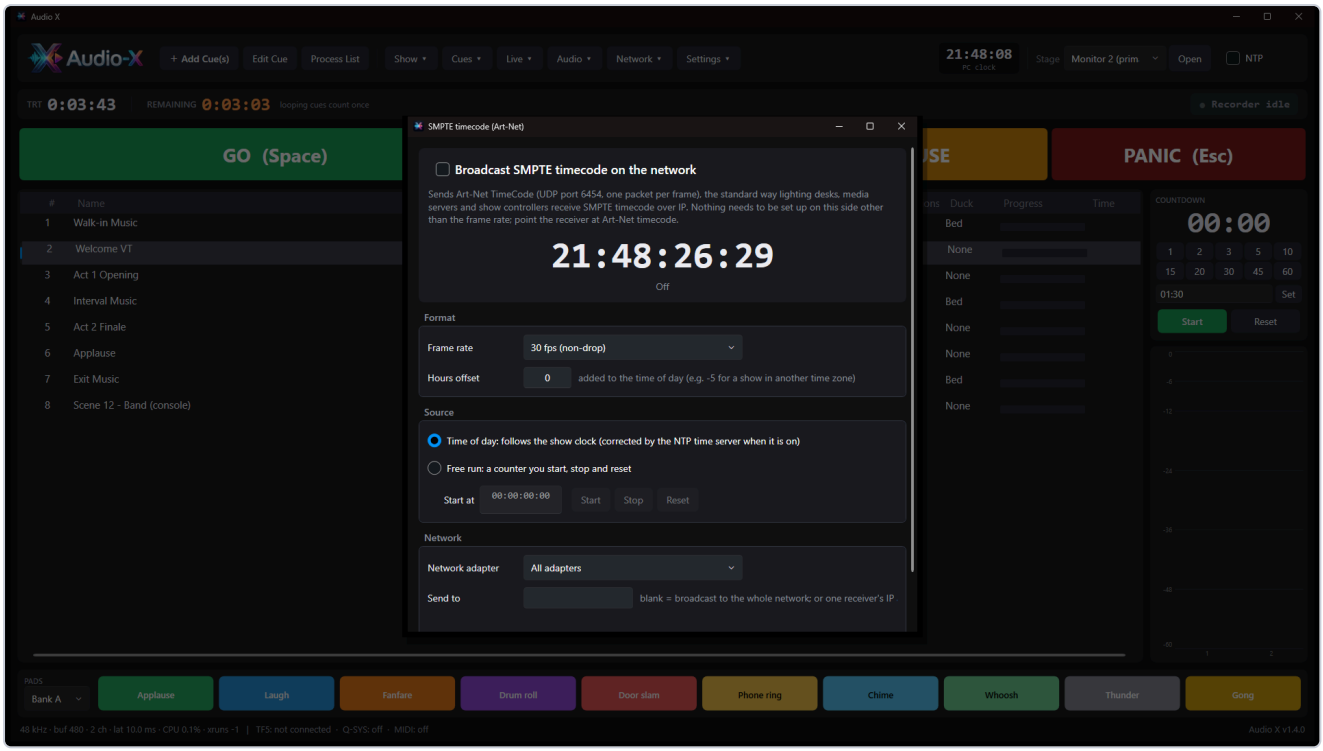
Shows the version, the copyright (C) 2026 Justin L Worley, the AGPL-3.0 licence notice and the no-warranty statement, the project web site and the contact address (contact@audiox.info). **Read the licence** opens the full text. See Licence and Copyright.

Time server (Network > Time server)



Tick *Set the clock from a network time server* to use a corrected time. The PC's own Windows clock is not changed. The corrected time is used by the stage display, its viewers, the radio's clock-time idents and the cue clock. Pick a server from the list (or type any host name), press **Test server**, then **Use this server**. For a live show choose a nearby, reliable server such as `time.cloudflare.com`, `time.google.com` or a pool.ntp.org region. The time is re-checked every 30 minutes. The **NTP** tick in the main window's top bar switches it on and off.

SMPTE timecode (Network > SMPTE timecode)



Sends **Art-Net TimeCode** (UDP port 6454) for lighting desks, media servers and show controllers.

- Frame rates 24, 25, 29.97 drop frame and 30.
- **Source**: the show clock (time of day, with an optional hours offset for another time zone) or a free-running counter with Start, Stop, Reset and a start value.
- **Network**: all adapters, one chosen adapter, or a single receiver's IP.
- The main window's clock shows the timecode being sent.

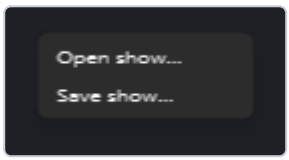
SMPTE LTC audio and MIDI timecode are not generated.

Menu Reference

Every item of the six menus, in the order they appear. Each item opens a window or does one thing; nothing appears in two menus.

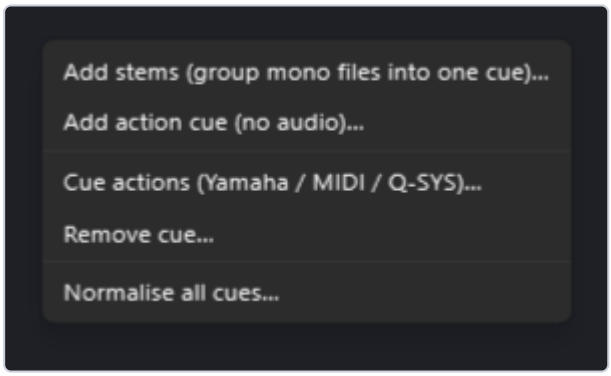
The top bar also has three buttons: **+ Add Cue(s)**, **Edit Cue** and **Process List**.

Show



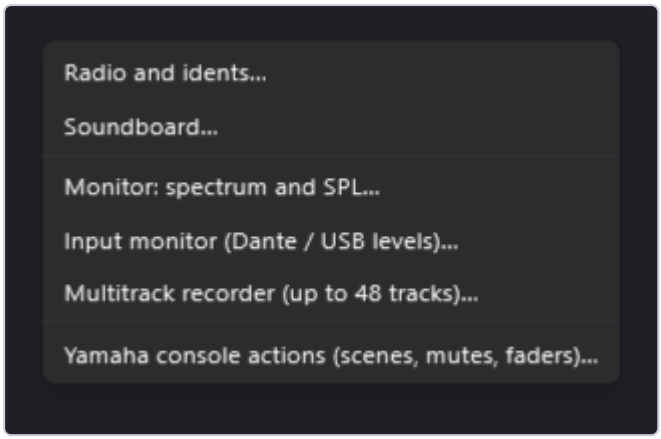
Item	Page
Open show	Cues and Shows
Save show	Cues and Shows

Cues



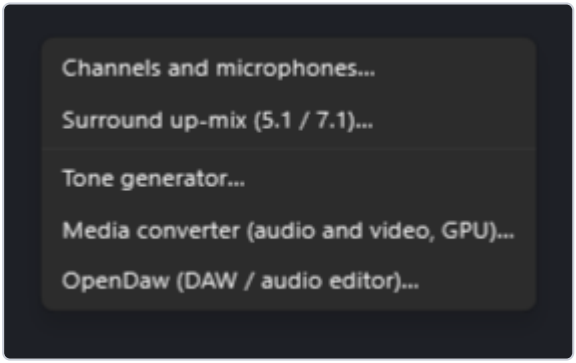
Item	Page
Add stems (group mono files into one cue)	Cues and Shows
Add action cue (no audio)	Show Control
Cue actions (Yamaha / MIDI / Q-SYS)	Show Control
Remove cue	Cues and Shows
Normalise all cues	Process List and Normalise

Live



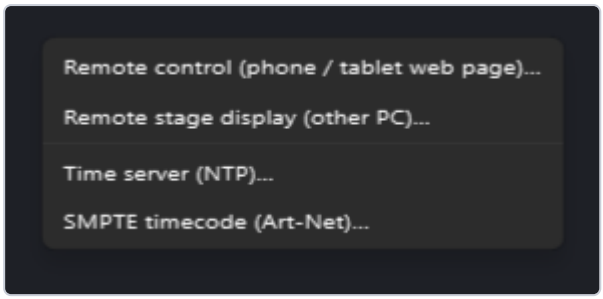
Item	Page
Radio and idents	Radio
Soundboard	Soundboard
Monitor: spectrum and SPL	Monitor: SPL and RTA
Input monitor (Dante / USB levels)	Input Monitor and Recorder
Multitrack recorder (up to 48 tracks)	Input Monitor and Recorder
Yamaha console actions (scenes, mutes, faders)	Show Control

Audio



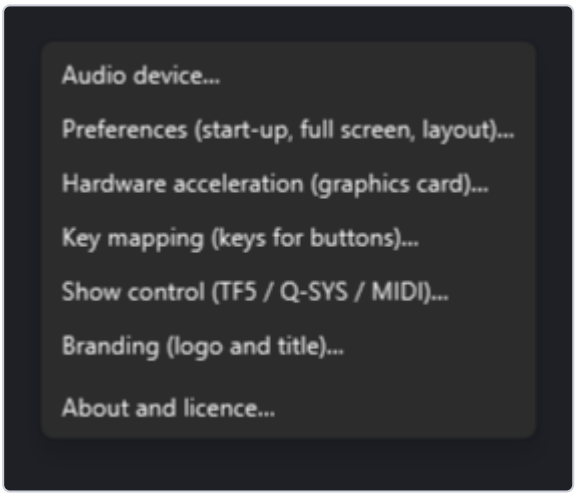
Item	Page
Channels and microphones	Microphones and Talkback
Surround up-mix (5.1 / 7.1)	Audio Tools
Tone generator	Audio Tools
Media converter (audio and video, GPU)	Media Converter
OpenDaw (DAW / audio editor)	Audio Tools

Network



Item	Page
Remote control (phone / tablet web page)	Remote Control and the Phone App
Remote stage display (other PC)	Stage Display
Time server (NTP)	Settings Reference
SMPTE timecode (Art-Net)	Settings Reference

Settings



Item	Page
Audio device	Getting Started
Preferences (start-up, full screen, layout)	Settings Reference
Hardware acceleration (graphics card)	Hardware Acceleration
Key mapping (keys for buttons)	Settings Reference
Show control (TF5 / Q-SYS / MIDI)	Show Control
Branding (logo and title)	Settings Reference
About and licence	Licence and Copyright

Keys

Key	Does

Space	GO
Esc	PANIC
F11	Full screen on and off

All other keys can be set in Key mapping.

Troubleshooting

No sound

- The status bar at the bottom of the main window should show the device, for example *48 kHz · buf 480 · 2 ch*. If it says no device is open, open it in **Settings > Audio device**.
- Check the cue's **Output** column: a cue plays where its Output says.
- Check **Gain dB** and the cue's trims. A cue with a start marker beyond the end plays nothing.

ASIO / Dante: "Device didn't start correctly"

Dante Virtual Soundcard needs a working wired Dante network before its driver starts. In the Dante Virtual Soundcard window:

1. The button at the bottom says **Stop** (it is running).
2. **Network Status** shows a speed (for example 1 Gbps) and **IP Address** shows an address. Wi-Fi, Bluetooth and bridged adapters are not supported.
3. **Audio Channels** is also the limit for Audio X.
4. The sample rate and bit depth are set in **Dante Controller**, not by Audio X: the rate in Audio X must match. The buffer size is set with the **Options** button in the Dante window.
5. Only one program can use an ASIO driver. Close any other program that has it open.

The Audio device window shows this checklist and the state of the PC's wired network adapters when the device will not start.

The announcement microphone does not work

Read the sentence under the level bar in the Radio window or on the phone page.

It says	Do this
The audio device is not open	Open it in Settings > Audio device.
Input N is not open on the audio device	Open more **Input channels** (ASIO only), or choose a PC input device instead.
Waiting for audio from the web device	On the phone, open the secure `https://...:8766` address, accept the certificate, allow the microphone and choose *This device's microphone*.
No signal / very quiet	Check the microphone is plugged in and not muted; check the right channel.

A phone cannot connect

- **Network > Remote control** must be switched on, and the PC and phone must be on the same network.
- Press **Allow in Windows Firewall** (both ports).

- Enter the PC's address with the port, for example `http://192.168.0.164:8765`. The Android app shows which of the two ports (plain and secure) answers.
- After five wrong PINs an address is locked out for one minute.

A Stage Viewer or browser cannot show the stage

- **Network > Remote stage display**: sharing must be ticked. Use its **Allow in Windows Firewall** button.
- A browser needs **Network > Remote control** switched on too, and asks for the web PIN.

The Media Converter or Process List says FFmpeg was not found

Install FFmpeg (for example `winget install Gyan.FFmpeg`) or press **Set ffmpeg** in the Media Converter and choose `ffmpeg.exe`. Audio X looks in the path you set, the Audio X folder, the system PATH and the WinGet links folder.

The GPU is not used

Open **Settings > Hardware acceleration**. If it lists no working GPU encoder, update the graphics driver and press **Look again**. Make sure the card is not set to *Off*. Only H.264, H.265 and AV1 video are encoded on the GPU; audio work runs on the CPU.

The Yamaha console does not respond

- Check the IP address in **Settings > Show control** and press **Test connection**.
- The status bar shows *TF5: not connected* until the first command. After a failed connection Audio X skips the console for five seconds, so a dead console cannot hold up the show.
- Use the RCP box in Show control to test a command line, and a *Yamaha raw* action for anything that differs on your console.

Levels of the radio look wrong when Spotify is the source

Spotify is captured with Windows process loopback. If the levels look uneven, switch off audio enhancements and spatial sound on the playback device.

Recording problems

- A recording needs input channels on the main device (ASIO).
- DROPOUTS means the disk stalled for longer than the buffer. Record to a faster disk.
- Close any other program that has the same ASIO driver open.

Where are my settings?

In `%AppData%\AudioX`. Shows are the `.axshow` files you saved. Deleting `settings.json` returns Audio X to its defaults.

Licence and Copyright

Audio X is copyright (C) 2026 Justin L Worley.

Audio X is free software. It is released under the **GNU Affero General Public License, version 3** (AGPL-3.0). This manual is covered by the same notice.

What the licence lets you do

- **Use it** for any purpose, including in a paid show or a commercial venue. There is no fee and no licence key.
- **Copy it and give it to others**, as long as you pass on the licence and the copyright notice with it.
- **Change it.** If you give your changed version to others, it must also be under the AGPL, and you must make its source code available to them.

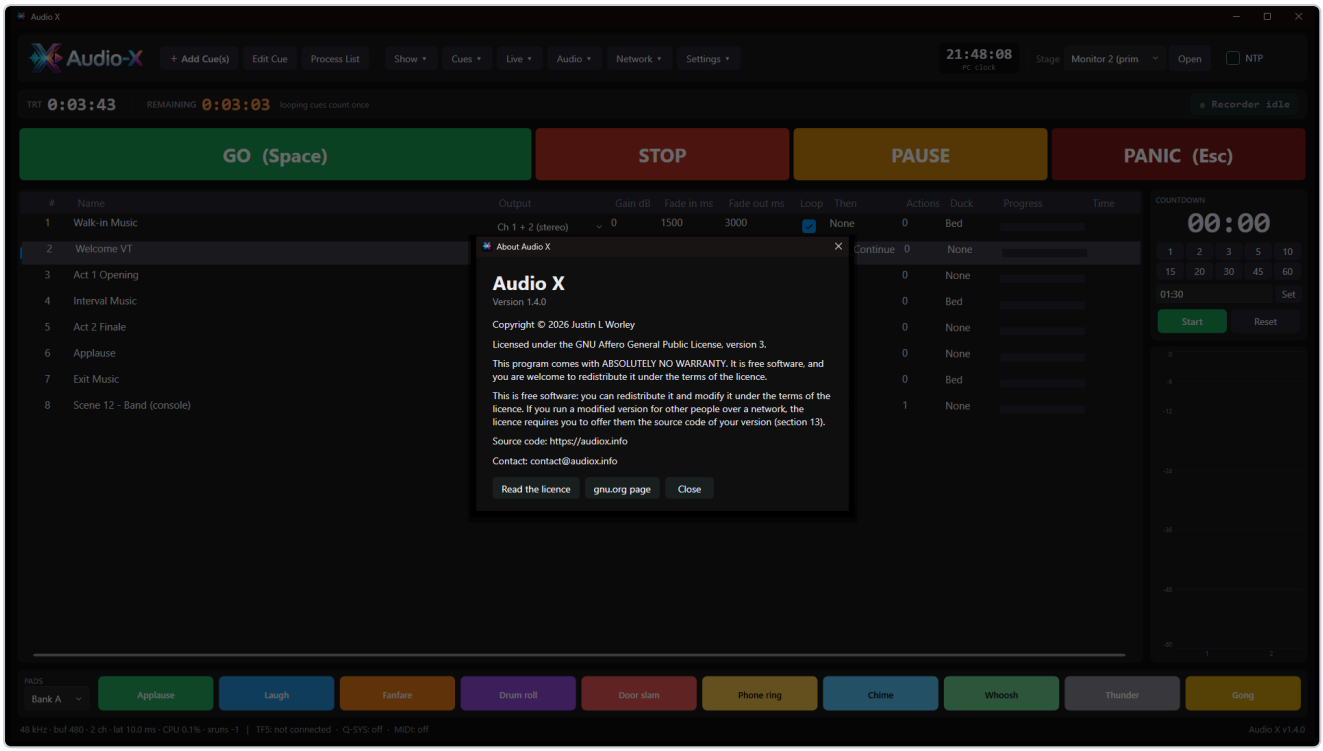
The network rule (section 13)

The AGPL has one rule that the ordinary GPL does not. If you **change** Audio X and let other people use your changed version over a network, you must offer those people the source code of your version. For Audio X this covers the phone and tablet web remote, the stage display and the Android app. If you only run the unchanged program, there is nothing to do.

No warranty

Audio X comes with **no warranty**, to the extent the law allows. The author is not liable for damage or loss of any kind arising from its use. In a live venue, rehearse every show and keep a backup way to play audio.

Where to find the notices



- **In the program:** Settings > **About and licence** shows the copyright, the licence and a button that opens the full licence text.
- **In the web remote:** the sign-in page and the Remote page have an **About and licence** link.
- **On the web:** audiox.info is the project web site.
- **In the installer:** the setup wizard shows the licence and asks you to accept it.
- **On disk:** `LICENSE.txt` (the full licence) and `NOTICE.txt` (copyright and third-party components) are installed next to `AudioX.exe`.
- **In the source:** every source file starts with a short licence header.

Source code

The project web site, **audiox.info**, has the downloads and the complete source code as a .zip file (`AudioX-1.4.0-source.zip`). `BUILDING.md` inside it explains how to build Audio X.

Contact

Questions, bug reports and ideas: **contact@audiox.info**.

Other software used

Audio X is built with JUCE, the Steinberg ASIO SDK, .NET and a few NuGet packages. Each stays under its own licence; the list is in `NOTICE.txt`. FFmpeg is not part of Audio X: you install it yourself and it has its own licence. Yamaha, Dante, Q-SYS, Spotify, NVIDIA, Intel and AMD are trademarks of their owners and are named only to describe what Audio X works with.