25-Seven Systems

AUDIO TIME MANAGER®
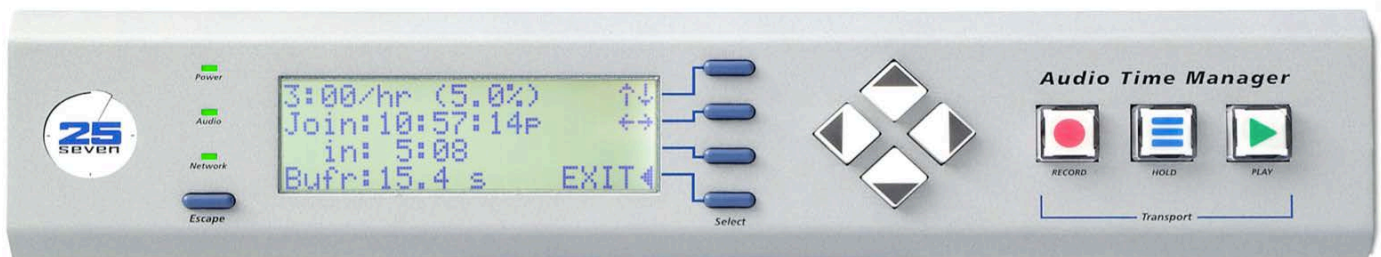
Operator's Manual

Version 2.2

THE TELOS ALLIANCE

It's About Time™

Warnings and Service Information

THE TELOS ALLIANCE™

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Table of Contents

Close-ups

Some sections of this manual explain the inner workings of the Audio Time Manager.
We've set them in a sans-serif face.
You don't need to read these sections to operate or install the system... but they are
interesting, and understanding them will help you use the system better.

About this Manual

These pages will help you get the most out of the Audio Time Manager™ (ATM).

- The first two pages describe ATM's basic concept and operation: if you're facing an ATM that's already installed in a rack, this may be all you need.

- The next two pages detail ATM's features and how they can benefit your operation.

- Then there's a detailed explanation of all the menus and controls, and instructions for setup and installation.

Audio Time Manager at a Glance

Think of the Audio Time Manager as a high-quality, rugged digital stereo recorder... with two important differences: it can record part of a show while playing back another part, and playback speed can be as much as 20% faster than the original program. In effect, this can create up to 12 minutes of "new" time per hour. It does this without pitch change, artifacts, or glitches, thanks to our proprietary processing algorithms; with most broadcast material, a 10% speed change—6 new minutes per hour—is virtually undetectable by most listeners. And the built-in Time/Rate Management Calculator gives operators complete control for smooth network joins, backtimes, and insertions with no special training or math. Audio Time Manager's simple interface reduces stress and the possibility of on-air mistakes.

On top of that, Audio Time Manager features a large pre-record buffer. Even if you get distracted and press the RECORD button late, you can "reach back" into the past and capture audio that occurred much earlier[1]. Our unique Cue Mode lets you change the point that a recording will start playing back from—earlier or later—*while you're still recording.* It always gives you another chance to make things perfect.

For more details on what ATM does and its special features, see page 7.

For information on how ATM's processing works, see page 23.

Basic 2-button Operation
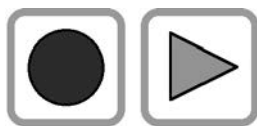
Audio Time Manager can give you complete control over time. It lets you delay the start of an incoming feed or recorded program but still have it end on time. It lets you "punch" breaks into a live event and catch up seamlessly. It can eliminate the need to back-time intros to network joins, so you can ad-lib and still join smoothly... even for events with random starts like news conferences. It can be automated, easily customized, and trimmed if conditions change while it's in operation. Complete instructions for these features start on page 8.

RECORD   PLAY

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[1] Up to an hour ago, depending on hardware configuration.

Or you can ignore these advanced functions and never read the complete instructions. If you can press two buttons, you can run an Audio Time Manager. Here are just a few applications:

Create time at the beginning of, or during, a network feed

When the network program starts or returns from a break, press RECORD on ATM. Run any local material—a weather forecast, breaking news, or additional spots. When you're finished with the local material, press PLAY on Audio Time Manager. That's it. The network program will start playing at a faster speed but with no pitch change or distortion, and then catch up with itself. Our unique Time/Rate Management Calculator™ (TRMC) figures out exactly how much speed change is needed—you don't have to do any math. It even tells you how long until the network is playing in real-time again.

Eliminate back-timing hassles

You've got 5 seconds before the network news comes down, but you've got 15 seconds of important copy left to read. You just need a little more time. Have ATM record the news at the top, finish your copy, then just press PLAY. Your listeners will hear your local copy and the complete newscast, and the newscast will end on time.

Smoothly introduce random-starting events like news conferences without awkward, time-wasting talk-ups

Ad-lib an introduction to, say, the Governor's live speech. Don't be afraid to finish a sentence or go to a promo or spot. When the speech actually starts, press RECORD. Complete your delivery, taking as long as you need. Introduce the event, then press PLAY. The feed will start from its beginning, slightly faster than normal, and then catch up with itself in real-time. It's as if they were waiting for your cue! You're still carrying the event "live", but you haven't wasted time awkwardly trying to anticipate the speaker's first words.

Create a custom break in the middle of a continuous program

Major breaking news has occurred and you're in the middle of a network show. Listen for any place to break in: the end of a sentence; a pause for a breath; the end of a song. Press RECORD, insert your news break, then press PLAY. Your listeners will hear the news break as well as the complete network show. Never again will you have to say "we return to programming already in progress"… now, the network waits for you.

Drop in IDs without stepping on the action

Easily add a station ID or announce a sponsor during a live remote or a ball game. Listen for a place to create a break, then press RECORD and insert your ID. When you're finished, press PLAY. The ongoing program picks up from exactly where you paused it, with nothing missing. It'll play slightly faster, then gradually slow down to real-time and join the ongoing feed smoothly. Our high-quality stereo signal path and proprietary algorithms mean that even music programs can be interrupted this way, without upsetting your listeners.

Run commercials during live events, without missing any action.

It's the big game and there's a time-out, so you go to a spot. But *whoops!*— they're playing again, and the commercial is still running. Don't worry… just press one button on ATM when they start up, and another when the commercial is over. Your listeners will hear everything, and you'll never have to say "well, folks, during the break you missed a great home run."

Add commentary in a hearing without disrupting it

You're carrying a live Congressional hearing with multiple speakers. Instead of interrupting and talking over testimony to identify them, use ATM to pause the live feed, announce the speaker, and resume. You'll catch back up to real time quickly, and your audience will hear your introductions and the complete testimony. Do this as many times as you need: you don't have to catch real time before you do your next insert.

Need easy-to-use time shifting?

ATM's perfect audio quality also makes it ideal for delaying programming without changing its running length. Time compression can be turned on or off without glitches, even during playback.

There's more to Audio Time Manager than a couple of buttons, of course, and plenty more cool things it can do[1]. Our Time/Rate Management Calculator™ lets you trim the playback speed while playing. It tells you exactly when ATM's output will drop back to real-time, and lets you trim that time as well. Our HOLD button lets you temporarily suspend time compression to keep fixed-duration events such as spots from being shortened.

Our unique Cue Mode lets you adjust when a recording started, while it's being recorded. You can move the start time later, to avoid coughs or awkward paper shuffling before an event actually started. Or you can move the start time earlier, *going back in time* to catch audio that happened before you pressed RECORD!

For example, you can **capture breaking news without even thinking about it.** Just patch your network alert channel or other news source into ATM when you're not using it for anything else. If news breaks and you miss it, just Cue back in time to the beginning of the news flash. Do an intro and play it on-air, or spool it into your newsroom system for later use.

Add another dimension of power by controlling ATM directly from your automation system. Use ATM to extend local breaks simply by adding them to your program log. Trigger ATM from your EAS equipment so that when emergencies like Amber Alerts occur, no regular programming is lost. Complete instructions begin on page 8 of this booklet.

Even if you do have ATM hooked up to your automation system, you can always take control manually and react to last-minute programming changes and breaking events.

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[1] **How do you use *your* ATM?** We want to know, so we can share techniques with other users. Drop us a note at info@25-seven.com.

Audio Time Manager Features and Benefits

Virtually imperceptible speed changes. At 10% faster—an additional 6 minutes per hour—most listeners won't even be aware ATM is in use. New technology makes it possible to do this with no choppiness, no stutters, no fuzziness, and absolutely no change in pitch or timbre. Even a 20% speed-up is glitch- and artifact- free… though at 20% with most sources, listeners will notice things being much faster than normal. Obviously, sources that speak slowly or music with a relaxed tempo can be sped up more than material that's already at breakneck speed.

No complicated programming needed. Two-button operation means you can react on-the-fly to unplanned events like weather alerts, late-starting press conferences, or holds in a concert or sporting event. Our Time/Rate Management Calculator™ (TRMC) adjusts to what *you* want to do.

Total flexibility with compression ratios. Change the upcoming playback ratio during record. Change it while playing. Or during playback, tell ATM when it should smoothly join real-time… and the proper ratio will be applied automatically. Change these ratios again and again, as many times as you want.

Unique Cue Mode for better on-air presentation. You can change the record-in point, after you've started recording! Move it later to clean up sloppy starts. *Move it earlier* to catch unexpected events that happened before you pressed RECORD!

Even advanced operations are simple. Most settings need just a few key presses. Softkeys mean that users don't have to memorize menu structures.

Keeps programs sounding natural. Some technologies can drastically cut into pauses, destroying a personality's delivery. Our proprietary algorithms intelligently compress tiny sections of soundwaves: pauses stay in proportion; pacing and inflection are kept.

Automatic ramp-down at end of compressed playback. No abrupt shifts: transitions back to real-time output are undetectable.

Works as an easy-to-use time shifter. You can delay programming by as little as a second or up to an hour, with or without time compression. Use simple push-button control, wire ATM to your automation system, or use both at the same time.

You're in complete control, all the time. Change vital parameters, even during compressed playback. Adjustments are smooth and glitch-free.

Flexible remote control. Run ATM from its front panel. Wire pushbuttons and LEDs to the built-in 8x8 GPIO. Connect using serial RS-232. Securely control ATM from a Web browser over your network!

Works with your satellite system. Advanced GPIO features let ATM lock incoming contact closures to specific incoming audio, and then fire outgoing closures when that audio plays out. Local liners, ID's and spots happen at the right point in the network program, whether you're using time compression or not.

Automatic creation of extra availabilities during network breaks. Special commands

let your traffic department put more spots in a break than would normally fit. ATM works with your satellite or network cue receiver and automation system to automatically fit those spots in, without an operator, and without losing any network content.

Time-Aware. ATM synchronizes itself to network clocks. Hook ATM up to any local data network that has Internet access, give it the address of a Network time Server (see page 30), and it'll always know the exact time. Or if your station has its own time server, lock ATM to it.

Quiet operation. The only moving part is a low-speed fan. Run ATM next to a live mic!

"Future-proof" audio quality. Excellent sound quality[1], both analog and digital stereo i/o, and PCM Linear processing mean you can use ATM now and keep using it tomorrow. No data reduction to mess up your sound. No distortion of the stereo field, even at extreme compression ratios. Time-compression algorithms are good enough to use on critical music programs.

Easy software upgrades over the Internet. We can help you connect your ATM directly to 25-Seven's servers to download the latest software. Future releases will even automatically check for updates.

Front Panel

Power Light   Audio Present   Select Softkeys   Cursor Buttons Record Hold   Play

Browser Active   Escape   LCD Screen   Transport Buttons
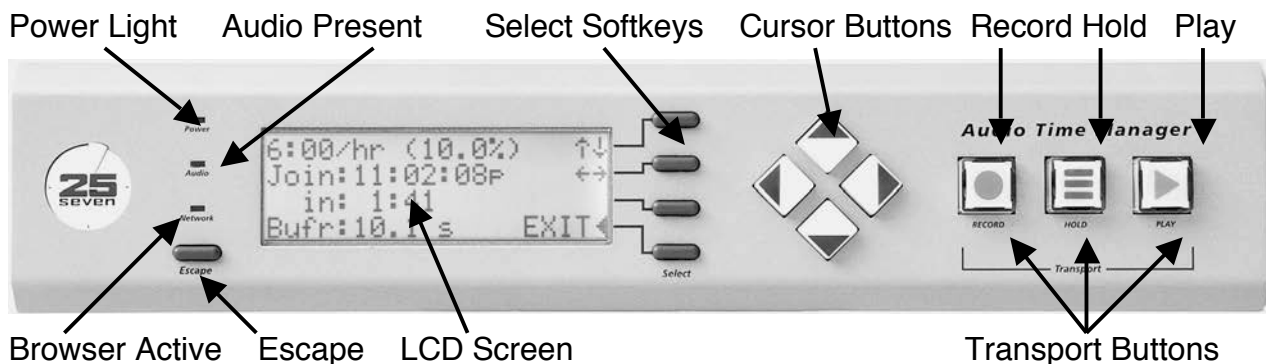
Details of the rear panel appear in the Installation section, page 35.

Detailed Operating Instructions

Manual modes

Audio Time Manager has four manually-controlled operating modes: Standby, Record, Play, and Hold. A fifth mode, Cue, is available during recording.

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[1] Signal/noise ratio = 84 dB with 10 dB headroom (94 dB dynamic range), A-weighted; THD @ 1 kHz = .01%; IMD (IHF) = .01%; frequency response = ±.5 dB 20 Hz – 20 kHz. Measured with analog i/o; expect slightly better results with digital i/o. No change in audio quality with increased time compression.

 (You probably thought we were going to try to weasel out of "excellent sound" in the footnote. Not us.)

Standby (Real-Time Play)

This is the default, when ATM powers up or after time compression has ended.

During Standby, the three transport buttons (RECORD, HOLD, PLAY) are **not** lit. Input signal (usually a network, satellite, or remote feed) is passed to the output in real-time, after a small processing delay. The processing delay—less than $^1/_4$ second—is necessary so that ATM can change modes instantly, without any audio glitches.

Even though the RECORD button isn't lit during Standby, ATM is constantly recording. We use a circular buffer to capture the most recent 30 minutes of audio[1], so you can go back and recall sounds that happened before you press RECORD. Learn how in the section on Cue Mode, page 13.

The LCD will look like this:
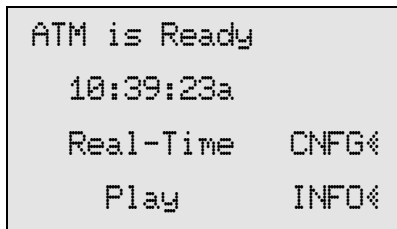
Status

Current time

Press the adjacent softkey to enter Configuration mode (page 25).

Press the softkey to read the current setup, IP address, software version, and similar information.

If the screen is different but no transport lights are lit, the system is in CNFG (Configuration) or INFO (Information) mode. It will still pass signal. You may press the ESCAPE button from these screens to return to the default Standby screen.

About softkeys

The ◈ symbol designates a *softkey*. There are four blue rectangular softkeys in a vertical column to the right of the LCD. When you see a ◈, you may press the softkey next to that symbol to activate the indicated function.

While ATM is recording or playing back, you will see ⬍ on the extreme right of the LCD. This is not a softkey; it tells you to use the UP or DOWN cursor buttons to change the time compression ratio.

While ATM is playing back or in hold mode, you will also see ◈◈ on the extreme right. This tells you to use the LEFT or RIGHT cursor buttons to change the join time.

Next step: Go from Standby or any other transport mode to Record. You can do this by pressing the RECORD button, or by using a remote control (page 21).
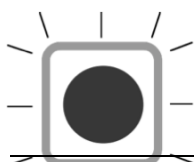
Record

Incoming signal is recorded to ATM's internal buffer. The output is muted to prevent mistakenly sending that signal to air.

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[1] Or more, depending on hardware configuration.

You can enter Record at any time by pressing the RECORD button. During Record, the button is lit.

During record, the LCD looks like this:

```
6:00/hr (10%)      ⇑⇓

Join: 6:24:36p CUE◁

  in:13:35    REC@◁

Bufr: 1:49.0  EXIT◁
```

Current ratio creates 6:00 minutes per hour, a 10% speed-up. Press the ⇑ or ⇓ Cursor buttons to raise or lower this ratio.
With the amount of material recorded and the specified ratio, system will play out and join real-time at 6:24:36pm. Read about Cue on page 13.
The Join time above will occur in 13 minutes 35 seconds. This display counts down. Press the softkey to see "Recorded at" time (below)
There is 1 minute 49 seconds of marked material presently in the record buffer. Press the softkey to stop recording and dump the buffer.

When you press the Rec@ softkey, the third line of the LCD changes to display the "Recorded at" time: the time-of-day when the current recording was started:

```
6:00/hr (10%)      ⇑⇓

Join: 6:24:36p CUE◁

Rec@: 6:22:19    IN◁

Bufr: 1:49.0  EXIT◁
```

The current recording started at 6:22:19pm. Press the softkey to return to the count down to Join time (above).

When you first press RECORD, the compression ratio displayed on the top line of the LCD will be the Default Ratio as set in ATM's Configuration Menu (page 27). This is the ratio that will be used when you press PLAY. But be aware that ATM is actually recording the incoming signal with no compression at all, for maximum quality and flexibility. Time compression happens on-the-fly during playback, and you can change the ratio at any time.

If you press the CUE◁ softkey, ATM continues recording and enters Cue Mode. This lets you adjust the starting point of the current recording. For more details, see page 13.

If you press the REC@◁ or IN◁ softkeys, the third line of the LCD toggles between how long it will take to join real-time, and the time this recording started.

If you press the EXIT◁ softkey or the ESCAPE button, ATM continues recording while showing you this confirmation screen:

```
WARNING: Dump audio

in the buffer?

          DUMP◁

          KEEP◁
```

Press the softkey to stop recording and erase the buffer.
Press the softkey to continue recording without interruption. This returns you to the record screen shown above.

If you press DUMP◁, ATM stops recording, dumps the buffer, and returns to Standby mode.

If you press KEEP◁ or the ESCAPE button, recording continues without interruption.

About DUMP and the Record Buffer.

We use terms like *dump* and *erase* for convenience. For most purposes, you can think of Dumping as emptying the buffer and erasing the recordings in it. That's what would happen if ATM were a conventional digital recorder.

In actuality, ATM is constantly recording to a circular data buffer, and Dump doesn't really discard audio. Instead, the oldest audio sample is constantly being replaced by the newest audio sample. So recent audio is always available—up to an hour, depending on hardware. Even if you "started recording" only a few seconds ago, you can still play audio that happened earlier.
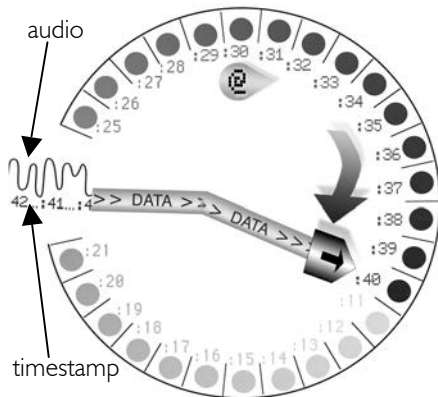

You can think of the buffer as being like the circle on the left, with slots to store tiny chunks of data around its edge. For this explanation, we'll assume an ATM with a half hour of storage, and that you pressed **RECORD** at 30 minutes past the hour.Incoming audio is constantly being marked with the current time. A pointer (the "record head") sweeps past the buffer's slots, making one complete circuit every half hour. The pointer stores the current audio and time data in the current slot, replacing any existing data there. In the drawing, the pointer is writing audio and time for 40 minutes past the hour.

If the pointer hasn't reached a slot on its current pass, that slot still holds data from the previous pass. That's why the next slot in the drawing above, just past the pointer, shows :11 minutes. With each pass taking 30 minutes, a slot can have audio and time data from up to a half hour ago.

The next thing our pointer will do is write data for :41 minutes (inset on right), replacing the data from :11 minutes. Then it will write :42, replacing :12, and so on.
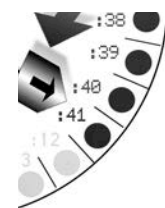
When you pressed the **RECORD** button, it put a "Recorded At" marker in the current slot: you can see it in the drawing at :30 past the hour. That's where playback will start from. Normally, that marker stays where you put it. But you can move it to any other slot in the buffer, earlier or later, to make playback start from a different time. For details, see Cueing on page 13.

By the way, even though our drawing looks a little like a clock, there's no noon or top of the hour. The pointer just keeps on going. Since the real time-of-day is stored with each sample, positions on an imaginary clock don't matter.

Also, we drew only a few dozen slots here, with a system storing data once per minute. ATM actually stores data for every audio sample of each audio channel—88,200 samples per second, or slightly more than 5 million per minute, almost 160 million in a 30 minute buffer.

About recording:

- The size of the record buffer is determined by each ATM's specific hardware configuration. All current systems (those shipped with version 2.0 or higher) hold one hour. Older ATM hardware may have less, but will hold at least 15 minutes of stereo material.

- ATM is *always* recording. Record Mode is actually a special state of the Time/Rate Management Calculator, rather than just an audio function. When you enter Record Mode, the T/RMC stores the current time as *Recorded At:* (Rec@), and starts calculating the Join time.

- When ATM has been in Record Mode long enough to make one complete pass through the circular buffer, it smoothly changes to Play mode. Of course, you can manually enter Play mode any time prior to that.

- When ATM enters Play mode, playback will start with the audio that originally occurred at the `Rec@` time.

- No time compression takes place during recording, only during playback. We did this so you can adjust the compression ratio while playing with no compromise in quality.

- When a recording starts to play, time compression will be at the ratio indicated on the top line of the LCD display. You can change this ratio while recording by tapping the UP or DOWN cursor buttons. You can also change compression ratio at any time during playback.

- You don't have to wait until the buffer has played out and you are in standby to press RECORD again. This means that even if you haven't caught up to real time, you can use ATM to do a new insert. Incoming audio is always appended to the end of the buffer, so content is never lost.

- The HOLD button temporarily turns off time compression, to protect the length of spots or so ATM can be used for time-shifting. It can be used during Record or Play. See the discussion of Hold on page 16.

Next step: You can stop recording, and start immediate playback, at any time by pressing the PLAY or HOLD buttons.

Or you can press the `Cue◁` softkey to enter Cue mode. This lets you change the `Rec@` time, and which audio will be heard when playback starts. See page 13.

About Record Advance and Cue Modes

We designed these features to help you capture events with random start times, and broadcast them with clean starts.

Imagine an aide introducing the Governor at a press conference[1]. Your talent is also doing an introduction as part of the commentary, so you don't need the aide's voice. You press RECORD just as the Governor gets to the podium. But Gov spends a moment shuffling papers and drinking water—sounds you don't want on the air. You could mute the podium mic, but that would leave an awkward silence or force you to fill with ad-libs. ATM gives you a better way.

Record Advance
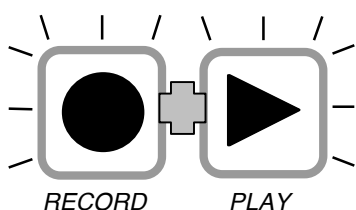
This feature updates the `Rec@` time to the current time. To use Record Advance, hold down the RECORD button and tap PLAY right before the real action starts.

RECORD   PLAY

[1] We're using a press conference for this example, but that's just for convenience. Record Advance and Cue work in stereo and with full fidelity. You can use them to cleanly broadcast the start of musical remotes or live performances without compromising sound quality.

The next time you Play or Hold, playback will start with whatever audio was occurring the moment you tapped PLAY.

In our example, you could start holding down RECORD any time during the paper shuffling, and tap PLAY when you see the Gov's mouth about to open. ATM will update the `Rec@` time and continue recording. When your talent finishes the opening commentary, press PLAY and the Governor will start cleanly.

- Pressing RECORD and PLAY *simultaneously* might not cause a Record Advance. You must be holding RECORD down *before* you tap PLAY. The RECORD button remains lit to confirm that you're recording.

- You can confirm that you've caused a Record Advance by checking the `Bufr:` display: It should reset to `0:00.0` each time you tap PLAY while holding RECORD.

- You can continue to hold RECORD and tap PLAY multiple times. Each time you do, `Rec@` is updated and the start of your recording is advanced.

Record Advance is a shortcut for pressing `EXIT◁`, then `DUMP◁`, then RECORD. It's usable at all times, no matter what mode ATM is in.

Because ATM's output is muted while recording, you may want to monitor a remote event's audio separately so you know when to use Record Advance.

Audible Cue Mode

`CUE◁` ⬭ This feature lets you Cue the `Rec@` marker to a new place, even after you've started a recording. In fact, the new `Rec@` can be at *any* audio that's in the buffer, even if you hadn't pressed record when the audio first occurred.

- Not sure when the Governor will say something worthwhile? Press RECORD when you think things are starting. After the recording starts, you can Cue forward to the first meaningful word.

- Pressed RECORD too late and missed the first question? Cue *backwards*, up to an hour earlier[1] (or less on earlier systems). Your recording will have audio that happened before you pressed RECORD!

- Missed the News Alert feed from your wire service, but know what time it started? Cue by time back to the beginning of the segment. The `Rec@` time displayed in Cue Mode is the actual time stamp of when audio was recorded. If ATM has been "listening" on your news channel, you can recover audio you might otherwise have missed.

To use Cue, you must be in Record mode, with the RECORD button lit. Then press the `CUE◁` softkey. The record button will start flashing, to let you know you're still recording.

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[1] New ATM systems include 1 hour of storage. Older hardware may have less. See page 11.

The LCD will change to look like this:
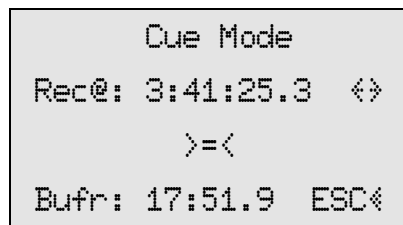
```
        Cue Mode

Rec@: 3:41:25.3   <>      Use the Left and Right cursor keys to change the Rec@ time.

        >=<              This line indicates speed and direction of cueing (see below).

Bufr: 17:51.9   ESC◁     Press the softkey to discard your changes and return to normal recording.
```

When you first enter Cue mode, the second line of the LCD shows the stored Rec@ time. The third line of the LCD will show >=<, indicating that Cue mode is paused.

Use the LEFT and RIGHT cursor keys to play or scan through audio in the buffer, and set a new Rec@ time. As you do, the second line of the LCD will show the time of day that the currently cueing audio was recorded. You can use this line of the LCD, in conjunction with the LEFT and RIGHT keys, to find an audio event that occurred at a specific time. Or you can cue by ear:

Tap LEFT, and audio starts playing in reverse. The third line of the LCD will show a reverse arrow icon (<=<).

Hold LEFT down to increase the scanning speed. Audio will start playing twice as fast as normal, in reverse. As you continue to hold LEFT down, the speed will double again (to four times as fast as normal), and again (to eight times), and so on. The arrow icon in the LCD will lengthen as the speed increases (<==<, then <===<, then <====<, and so on).

Tapping RIGHT will make audio cue forward. Holding RIGHT down will double the forward scanning speed, then double it again, and so on; the LCD will display a forward arrow icon (>=>) that lengthens with the speed.

You can pause Cueing at any time by pressing both LEFT and RIGHT at the same time.

- When you've found the audio you'd like the recording to begin with, you can tap RECORD to store the new Rec@ and return to normal recording. When you eventually enter Play Mode, ATM will start playing from this new Rec@ time.

- Or you can tap PLAY now, to start normal time-compressed playback from this point.

- Or tap HOLD now, to start uncompressed playback from this point (see page 16).

- You can tap these transport buttons while audio is either cueing or paused.

- To discard your changes and return to recording with the original Rec@ time, press the ESC◁ softkey.

Cue's time limitations

While ATM uses a circular buffer, it won't let you cue 'around the clock': Rec@ can't be set earlier than the current time-of-day, minus the length of the buffer itself. This keeps you from inadvertently overlapping the record pointer, and hearing audio out of sequence.

If you have recently powered ATM up, it might not have been on long enough to completely fill the buffer. In this case, the system won't let you cue earlier than there is valid audio.

And of course, you can't cue any later than the current time-of-day. That would require predicting what audio will occur in the future. (We're good, but not *that* good.)

Play

Audio in the buffer is sent to ATM's output, sped up by the time compression ratio. This ratio can be changed during playback. As the buffer nears empty, compression is gradually ramped to zero for a smooth transition to real-time. When the buffer is completely empty, playback stops, and the system returns to Standby Mode and the bottom two lines of the LCD display `Real-Time Play`.

You can enter Play whenever there's material in the buffer by pressing the PLAY button. ATM will also automatically start playing when the buffer is full. Whenever ATM is in Play mode, the PLAY button will be lit.

During play, the LCD looks like this:

```
6:00/hr (10%)       ↑↓

Join: 6:24:36p      ‹›

  In: 4:58:00 REC@‹

Bufr: 1:49.0  EXIT‹
```

Current ratio creates 6:00 minutes per hour, a 10% speed-up. Press the ↑ or ↓ Cursor buttons to raise or lower this ratio.

At the specified ratio, system will join real-time at 6:24:36pm. You may change this by using the ‹ or › Cursor buttons.

The Join time above will occur in 4 minutes 58 seconds. Press the softkey to display the time when this recording started.

There is 1 minute 49 seconds of material presently in the record buffer. Press the softkey to stop playback and dump the buffer.

Pressing the third softkey toggles the third line of the display between a countdown to the join time (`In:`), and the `Rec@` time of this recording.

The `Join:` and `in:` displays show when the buffer will empty and ATM will smoothly revert to Standby mode. If you've recorded silence following the feed, ATM won't join until that silence has played out. So it's a good idea to always take your cue from the playback audio, not from the `Join:` time.

Stopping playback: Playback normally continues until the Join time; then the system smoothly switches from Play to Standby and passes the input signal to the output. You may manually stop playback at any time by pressing RECORD; this mutes the output and the system goes smoothly into Record mode.

• ATM normally plays until the buffer is empty. If you press RECORD during playback, there will still be material in the buffer. *This older audio is kept in the buffer* and will be heard at the start of the next playback. New material is added to the end of the buffer, and will play when the older audio is finished.

• If you want to empty the buffer and start recording fresh—so that the new material will be the only thing that plays—use the Record Advance function described in the previous section (hold RECORD while you tap PLAY).

You may also stop playback and empty the buffer by pressing the `EXIT‹` softkey or ESCAPE. Playback **continues** while you're given a confirmation screen: from this

screen, you may press the KEEP⁴ softkey or press ESCAPE to continue playback, or press the DUMP⁴ softkey to empty the buffer and return to Standby.

"Phantom Recording"

Even though the **RECORD** button is not lit during Play mode, ATM is still recording to the buffer. It will play this audio back, compressed, during the "catch up" to real-time. Because of this feature, no incoming material is ever lost.

You may notice the Bufr: indicator counting down during play… that's because the buffer is playing out old audio faster than it's recording new audio, not because ATM isn't recording.

About playback

Default ratio. Playback starts with the compression ratio shown on the screen. When you start recording, this will be the default ratio set in ATM's CNFG⁴ mode (see next paragraph). You can change the ratio any time during record or playback with the ⇑⇓ cursor buttons. You can also change the ratio during playback by adjusting the Join: time with the ⇔ cursor buttons. When you do the latter, ATM's Time/Rate Calculator changes the compression ratio so you join real-time at exactly the specified time.

You can change the default compression ratio only while ATM is in Standby mode. Press CNFG⁴ and then AUDIO⁴. Then press NEXT⁴ a few times until you see Default Comp:. Use the Up and Down cursor buttons to adjust the value, press DONE⁴ to save the change, and ESCAPE to return to Standby.

Ramp time. As the buffer gets close to being empty, ATM automatically starts reducing the compression ratio so there is no compression the moment before ATM switches back to Standby Mode and Real-Time Play. This prevents annoying and abrupt tempo shifts and provides a smooth and imperceptible transition to Standby mode. You may notice more rapid screen updates for Join Time or compression ratio as ATM's playback gets close to real-time. This is normal behavior indicating that our Time/Rate Management Calculator is hard at work.

You can adjust how long this ramping takes while ATM is in Standby mode. Press CNFG⁴ and then AUDIO⁴. Then press NEXT⁴ a few times until you see Ramp Time:. Use the Up and Down cursor buttons to adjust the value, press DONE⁴ to save the change, and ESCAPE to return to Standby. You can not adjust this value when ATM is recording, holding, or playing.

Hold

Hold puts the ATM in play mode with no time compression. Whatever is in the buffer plays out at its original rate. Hold makes it easy to respect the length of recorded spots or breaks while playing, or to use ATM for basic time-shifting.

You may enter Hold mode any time you are recording or playing, by pressing the HOLD button.

During Hold, the LCD will look like this:

```
6:00/hr (10%)     ↑↓

Join: 6:24:36p    ◁▷

Holding:0:30

Bufr: 1:49.0  EXIT◁
```

When hold times out, play will be at a 10% speed-up. Press the ↑ or ↓ Cursor buttons to raise or lower this ratio.

At the specified ratio, system will join real-time at 6:24:36pm. You may change this by using the ◁ or ▷ Cursor buttons.

This counts down amount of time left to hold, before returning to Play mode.

There is 1 minute 49 seconds of material presently in the record buffer. Press the softkey to stop playback and dump the buffer.

Each tap of the HOLD button adds to the `Holding:` time by a preset amount. Changing the `Holding:` time also changes the `Join:`, since less time is now available for compressed playback.

The Hold Time preset amount is factory set to 30 seconds but can be adjusted when ATM is in Standby mode. Press `CNFG◁` and then `AUDIO◁`. Then press `NEXT◁` a few times until you see `Hold Time:`. Use the Up and Down cursor buttons to adjust the value between 1 and 120 seconds, press `DONE◁` to save the change, and ESCAPE to return to Standby.

For these descriptions, we'll assume you've left the Hold Time at 30 seconds:

- As soon as you press HOLD, the `holding:` timer will start counting down from 0:30 seconds.

- If you tap it twice in rapid succession, the `Holding:` timer will count down from `1:00`; three quick taps will give you `1:30`, and so on.

- If you tap HOLD while the timer is counting down, it adds 30 seconds to the current hold time. (For example, if `Holding:` reads `1:15` when you tap it, the timer will immediately jump to `1:45`.)

- When the timer reaches `:00`, ATM automatically goes to time-compressed Play mode. This happens no matter what mode it was in when you first pressed HOLD.

- During Hold, new audio is being recorded at the same rate old audio is being played. So even though ATM is recording and playing at this time, the `bufr:` display doesn't change.

You may cancel Hold at any time by pressing RECORD or PLAY. This puts ATM in the selected mode, and zeros the `holding:` timer.

Using Hold while playing

If you've recorded a program and are now playing it back, you can temporarily suspend time compression by pressing HOLD. While the HOLD button is lit, playback continues from the buffer but at the original speed.

- Use this feature to give network advertisers their full money's worth for a :30 or :60 spot.

- Or, use this feature when replacing a network break with a local one. ATM will make sure the break lasts its original length.

- For testing purposes, you can toggle between Hold and Play to quickly compare how something sounds at various time compression rates vs. the original, uncompressed source.

Obviously, Hold mode can be used for other length breaks and isn't restricted to spots.

In the graphic examples that follow, we'll use `Courier` to represent normal speed audio and `Courier Condensed` for time-compressed audio. The timings, compression ratios, and hold times are arbitrary.

Example of Hold from Playback		
ATM Buffer (already recorded)	Operation:	ATM Output

`This is Talkfest. Today's guest is...`
 	
`(8:00 later)...we'll be right back after a short break.`
 	
`This is the Neural Network.`
 	
`This is the year for you to drive a brand-new Phud. Your Phud dealer has lots of deals...`
 	
(30 seconds later) *`That's Phud, when you need to Get There.`*
 	
`Welcome back to the Neural Network's Talkfest...`
	Operator presses PLAY
	
	Operator hears cue, presses HOLD
	
	Hold automatically times out or operator presses PLAY
		`This is talkfest. Today's guest is...`
		
		`(7:00 later)...we'll be right back after a short break.`
		
		`This is the Neural Network.`
		
		`This is the year for you to drive a brand-new Phud. Your Phud dealer has lots of deals...`
		
		(30 seconds later) *`That's Phud, when you need to Get There.`*
		
		`Welcome back to the Neural Network's Talkfest...`

Using Hold while recording

When the RECORD button is lit, ATM's output is muted. If you then press HOLD, ATM's output turns on and Play mode starts from the top of the buffer, with no time compression. The signal is essentially time-delayed by however long ATM had been recording before you pressed HOLD.

At the same time, ATM continues recording the incoming signal, storing it in the buffer for later playback. Since the buffer is playing with zero compression and simultaneously recording in real-time, the `Bufr:` display doesn't count up or down.

When Hold times out, ATM smoothly switches to time-compressed playback, and finishes playing what's in the buffer. You can also force this switch to time-compressed playback by pressing PLAY at any time. Or you can press RECORD from Hold mode, muting the output and continuing to write incoming material to the buffer.

Example of Record to Hold to Play		
ATM Input	Operation:	ATM Output

`This is Talkfest. Today's`
`guest is...`
 	
`(:30 later)...we'll be`
`right back after this news`
`break.`
 	
`In the headlines today, a`
`Boston high-tech company`
`has come up with a way for`
`broadcasters to regulate`
`time...`
	Operator presses RECORD
	
	Operator presses HOLD
	
	Hold times out or operator presses PLAY
		(Muted while you run local content)
		
		`This is Talkfest. To-`
		`day's guest is...`
		`(:30 later)...we'll be`
		`right back after this`
		`news break.`
		`In the headlines today, a Boston high-`
		`tech company has come up with a way for`
		`broadcasters to regulate time...`

Example of Record to Hold to Record		
ATM Input	Operation:	ATM Output

`This is Talkfest. Today's`
`guest is...`
 	
`(:30 later)...we'll be`
`right back after this news`
`break.`
`In the headlines today, a`
`Boston high-tech company`
`has come up with a way for`
`broadcasters to regulate`
`time...`
	Operator presses RECORD
	
	Operator presses HOLD
	
	Operator presses RECORD
	
	Some time later, operator presses PLAY
		(Muted while you run local tease)
		
		`This is Talkfest. To-`
		`day's guest is...`
		`(:30 later)...we'll be`
		`right back after this`
		`news break.`
		(Muted while you run local news brief)
		`In the headlines today, a Boston high-`
		`tech company has come up with a way for`
		`broadcasters to regulate time...`

Delay Mode for time shifting

At any time during Record, Play, or Hold you can put ATM into Delay mode: audio passes through the system unchanged, except it's delayed by the length of time between when you pressed RECORD and when you press HOLD or PLAY. To enter delay

mode, press the DOWN cursor button or tap it multiple times until the compression ratio reads `0.0%`.

During Delay mode, the second line of the LCD reads `Join:Infinite`. When playing in Delay mode, the `bufr:` display freezes because you are playing at exactly the same rate as you are recording new material[1]. During Delay mode playback, `bufr:` indicates the precise length of the delay.

You can get out of Delay mode by tapping the UP cursor button and setting a new compression ratio. You can also exit Delay mode by pressing ESCAPE or the `EXIT◁` softkey, and dumping audio. The next time you record, ATM will revert to the Default compression ratio.

- If you want to repeatedly use ATM as a delay, without time compression or automatic join features, set audio `default comp:` to `0.0%` (page 27). Operators will still be able to use time compression, by tapping the UP cursor button as explained in the previous paragraph.

- If want to use ATM as a delay that's triggered by network commands that might not follow a strict schedule, set `Hold Time` to `Time Shift` (page 29). This lets ATM continue to time shift, after the HOLD button has been pressed, until the PLAY button is pressed. Then connect your automation system to ATM's GPIO triggers for Hold and Play (page 30).

Front-panel indicators

➢ **Power LED** Nothing special. It lights when AC is supplied to the rear panel power jack. ATM is designed for 24/7[2] operation and does not have an on/off switch.

➢ **Audio LED** When using analog inputs, this lights when the input level is equivalent to –30 dBFS or higher; in most installations this will be –8 dBu or –12 VU. See page 27 for a discussion of how dBFS and dBu can relate to each other.

   When digital input is selected as the audio source, this LED lights when ATM's digital input is locked to a valid signal. On Axia /Livewire™ configurations, this will lights when ATM successfully connects to a destination address.

➢ **Network LED** This lights when ATM is being controlled from a Web browser over the network.

---

[1] Yes, even though you're in Play mode, ATM is still recording. See "Phantom Recording" on page 16.

[2] Not our trademark. We're _25_-seven.

Remote control

ATM supports three different remote control types, for maximum flexibility. All three types can be used simultaneously.

Parallel (GPIO or "Contact Closure") Control

There are eight inputs and eight outputs for remote control on a rear panel DB-25 connector. Each can be assigned to ATM functions through a configuration screen, described on page 27. Multiple inputs or outputs can be assigned to the same function if you need isolated circuits. Inputs and outputs are opto-isolated, active low; wiring details are on page 36.

Serial (RS-232) Control

There is a DB-9 serial port on the rear panel for remote control, which can be connected to a computer via a null modem or to various 3rd-party control devices. ATM uses a simple ASCII-based command language; communication and language specs are on page 40. Wiring details for the connector are on page 36.

Web Browser based Control

An RJ-45 jack on the rear panel lets you connect ATM to your local area network via 10/100-BaseT Ethernet. ATM includes an internal password-protected Web server; properly-configured browsers on the network can log into the server for an on-screen emulation of ATM's front panel. This same Ethernet port can be used to automatically update ATM's internal time-of-day clock and will be able to log into 25-Seven's website for future software updates. Configuration details are on page 29.

Log-On Instructions

Once ATM is properly configured and connected to a network, you can log into its internal Web server by using its IP address and unit serial number. To find this information, press the `INFO‹` softkey from the standby screen. Then tap `NEXT‹` until you see `Network Addr:` on the top line of the LCD. The four numbers on the second line, separated by periods, are the IP address.

```
Network Addr: NEXT‹

192.168.1.102 PREV‹

Mask:

255.255.255.0 DONE‹
```

For this installation, ATM's IP address is 192.168.1.102. The number will probably be different at your station.

Subnet Mask determines the zones in your network. Check with your IT manager to make sure this number is correct for your setup...

...but don't worry about entering it as part of the IP address.

Press `NEXT‹` again to view the serial number (`SN`) of the unit. You will need this number as part of the password.

Launch a Web browser[1] from a computer on your local network, enter ATM's IP address in the browser's Address or Location field, and click Go or press ENTER on the computer.

A password confirmation screen will come up. Use "`atmweb`" for the user name, and "`atmwebxxx`" for the password, where *xxx* is your unit's serial number. Note that these passwords must be lower-case.

Once you connect , you should see a version of ATM's front panel on your computer screen.

If ATM is connected to your local network, you should be able to access this screen from a web browser once you login. The screen works just like ATM's real front panel.
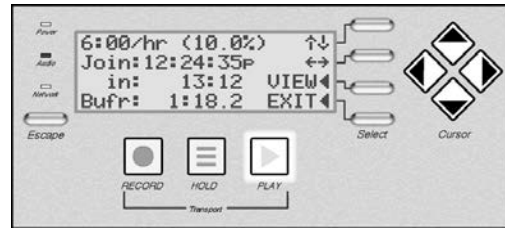
- You can click the on-screen transport controls, cursor buttons, softkeys, and ESCAPE to control ATM, just as you would from ATM's front panel. The on-screen LCD will update to reflect changes on ATM's front panel LCD, and on screen buttons will light reflecting the state of the transport.

- Once you've clicked an on-screen control to initialize the interface, you can use your computer's main keyboard or numeric keypad to control ATM:

ATM function	Press on main keyboard	or, press on numeric keypad
Record	R	0
Hold	H	. (decimal point)
Play	P	Enter key
Softkeys 1 through 4	1 through 4	1 through 4
Escape	E or Escape key	- (minus)

Note that the ATM screen graphic must be the active window in your computer for these keyboard shortcuts to work. If you click your mouse on another web page in your browser, or on another application, you'll disable the shortcuts.

- Web-based control relies on Macromedia Flash being installed in your browser[2]. Flash communicates with ATM over port 5444. If this port is blocked by your firewall, Web-based control will not work. You will see `Connect Failed` on your browser, and the system will try to connect again.

- The password login is meant as a safety mechanism against unintended tampering with ATM's controls. Shutting the Web browser down after you are finished will ensure that the next user must login before using the Web interface again.

---

[1] Browser must be configured with Macromedia Flash (version 9 or higher). If you aren't sure if a browser has Flash installed, try connecting anyway: ATM's internal Web server will test for the plug-in, and provide a link to a free download if it isn't installed.

[2] When using recent versions of Microsoft Internet Explorer for Windows, you may find that your first mouse click on the ATM Web Interface is ignored. Due to a patent lawsuit, Microsoft has decided to require that Flash applets on web pages be "activated" before use. Once you've activated the applet (by clicking on it), it will function as long as the web page is open.

When using the Web browser to control an ATM in another studio, always check to make sure the unit is not currently in use! Remember that an ATM in standby mode may actually be feeding real time audio to your program bus.

How ATM's Time Compression Works

ATM shortens audio segments without distortion, glitches, or pitch changes.

The basic concepts for time compression were developed a half century ago. But both the science and the art have improved dramatically over the decades, due in large part to intellectual contributions from people working with 25-Seven. Our continuing efforts have resulted in sophisticated algorithms that, when properly applied, are virtually undetectable.

While our software is complicated, the basic principles are easy to understand. Imagine a sound, in this case an announcer reading copy:
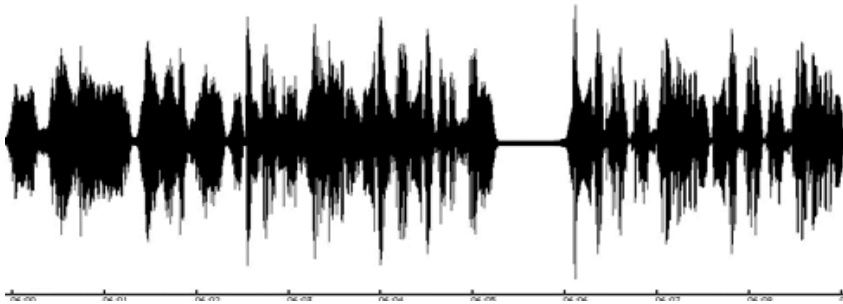
An unprocessed announcer. The ticks along the bottom are .01 second apart.

If you want this sound to take less time, you have to make each wave faster. That's how vari-speed on a tape recorder works. ATM doesn't have tape, but does a similar thing by adding a second pointer to the circular buffer[1].

While the record pointer (grayed in the drawing) is recording new audio at a constant speed, filling the buffer in real-time, the playback pointer is moving through the buffer *faster* than real-time[2]. The actual speed of the playback pointer is determined by the compression ratio.

Unfortunately, faster waves have a higher pitch. That's why vari-speeding a piece of music throws it into a new key. If we did that to the announcer, we'd get something that sounds like a chipmunk.

---

[1] You can review the Circular Buffer concept on page 10.

[2] Even though the playback pointer is moving faster than the record pointer, it'll never overtake it. As the playback pointer gets close to the record one, it slows down (*ramping,* page 16). When it reaches the same time as the record pointer, ATM joins real-time and goes into standby mode.
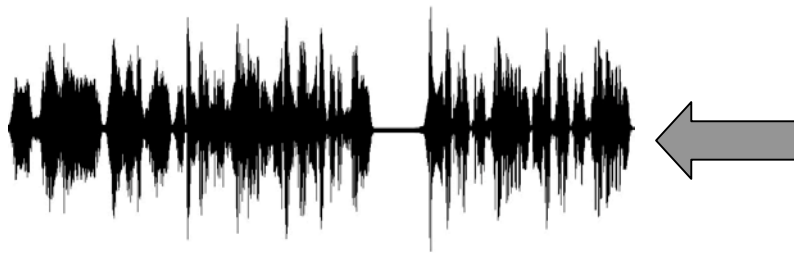
You could make things faster by squeezing the waves together, but that would change the pitch.

An approach used by some systems is to look for pauses—places where the volume falls below a pre-set threshold—and cut those pauses out. We don't do that.

You could also speed things up by leaving waves the same length but cutting out the pauses. We think that just sounds bad, and destroys the talent's delivery.
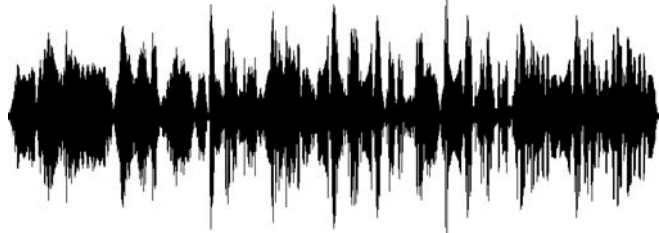
The problem is, deleting pauses can completely destroy the pacing and delivery that your air talent works so hard to present. Since music rarely has pauses, it would also be impossible to effectively speed up music this way.

We invented a better kind of time compression

ATM squeezes time imperceptibly by working on a much tinier scale—not syllables, words, and pauses, but individual waves. Let's zoom in on that sound:
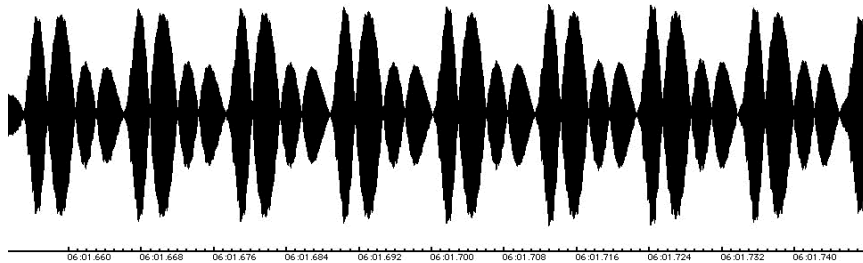
Zooming in on the same sound. The ticks are now .001 second apart.

Vowels like the one shown above, and sustained musical notes, consist of repeating patterns of waves. ATM recognizes these patterns, and—using artificial intelligence—deletes just enough repetitions to make the sound faster without making it sound different. Each individual wave still takes the same amount of time, so the pitch doesn't change, but there are slightly fewer waves overall.

Think of ATM as making tiny, very precise splices in a long continuous sound. This kind of editing could never be done in an audio workstation or with a razor blade.

Some sounds, including most consonants and almost all percussion instruments, don't have nicely repeating waves.

Consonants don't have nicely repeating waves, so ATM takes a different approach.

ATM recognizes these sounds, and applies psychoacoustic principles to make cuts only where the ear isn't likely to hear a change. A few kinds of sound, such as the short consonant "T" or the first part of a cymbal crash, don't get cut at all.

Thanks to a powerful computing engine, ATM can apply this kind of intelligent analysis and manipulation thousands of times per second. Pauses are analyzed and processed the same way, so everything is kept in proportion and details of breath and pacing are never lost. The result is smooth, natural sounding time compression that, in many cases, is totally undetectable to the listener.

Of course some program material, such as already sped-up commercial disclaimers or very fast-paced music, might become unnaturally fast—particularly if ATM is set for a very high compression ratio—and listeners *will* spot that. In extreme cases, some splices may even be audible. But used intelligently, ATM can become a regular part of your station's sound. It can also impart a sense of excitement and urgency: research shows that listeners have a preference for compressed normal speech, rather than rapid talking.

The technical details

We use variable-width analysis windows to simultaneously check the pitch, tempo, and spectral characteristics of the audio. Additional algorithms are applied to preserve musical rhythms and stereo placement. Our software is the product of years of research into the advantages and pitfalls of other time-compression algorithms.

How'd we come up with such clever solutions? Our algorithms were invented by Dr. Barry Blesser, who heads our audio programming team. He's known as "The Father of Digital Audio Signal Processing", and created the first successful digital reverb in 1976. In the years since, Dr. Blesser has amassed a reputation for technologies from audio workstations to handwriting recognition. A former M.I.T. professor, he served as Chief Technology Officer for Studer and is a Fellow and Past President of the Audio Engineering Society.

System Settings

Configuration

You can tailor Audio Time Manager's behavior to fit your station's operations, by using the Configuration Screens. These let you adjust audio settings, time compression, network connections, remote control, and system settings such as the internal time-of-day clock (if you're not using a Network Time Server) and screen brightness.

Configuration is only available while the ATM is in Standby mode and showing `Real-Time Play`. Press the `CNFG` softkey to enter the Configuration menu. There you can select which aspect of the ATM you want to change, and then cycle through individual Configuration screens to set specific characteristics. The menu structure is detailed below.

- If any of the transport buttons are lit, you cannot change the Configuration.
- If you are in any of the Configuration screens, you cannot Record. If a situation arises where you must immediately record, either:
    - press ESCAPE to leave the Configuration screen without saving changes, or
    - press `DONE` to leave the Configuration screen and save your current changes.

Then you can press RECORD. It is not necessary to return all the way to the `Real-Time Play` screen to enter Record mode.

All Configuration operations work the same way:

- When you press the `CNFG` softkey, you'll see a screen like this:

Select	AUDIO
config.	NET
category	GPIO
	SYS

Includes audio i/o settings and time compression default values.

Includes settings for use with Ethernet: DHCP and IP addresses.

Lets you select which ATM states are activated or reported by the 8 general purpose input/output connections.

Sets clock and LCD options.

- Then select a category by pressing the `AUDIO`, `NET`, `GPIO`, or `SYS` softkeys.

- When you do, you'll see the first screen in that category, with an identification of what characteristic it controls and what the current value is. Here is a typical one:

Digital Out:	NEXT
AES/EBU	PREV
↑↓ Chg value	
	DONE

This screen lets you set the Digital Output format. When you're done, press the softkey to go to the next screen.

Current value is AES/EBU. Press the softkey to go to the previous screen.

Use the Up and Down Cursor buttons to change the current value

Press the softkey when you have made all the changes in this category.

- The value which can be changed will flash[1]. In this case, `AES/EBU` can be changed to `s/pdif` by pressing the Up or Down cursor, so `AES/EBU` is flashing.

- Pressing the Up or Down cursor always changes the flashing value. Some screens have multiple fields that can be changed; in these screens, pressing the Left or Right cursor changes the active field.

- When you are satisfied with the value, or if you don't want to change that specific characteristic, press `NEXT` to move to the next screen. Press `PREV` to go back to the previous screen.

- Pressing `NEXT` or `PREV` temporarily applies the changes you have made on that screen, and they remain in effect while you stay in that category.

- When you have made all the adjustments you want in a category, press `DONE`. This saves all that category's changes at once.

- If you want to discard all the changes made since you entered a specific category, and revert to the previous values, press ESCAPE.

- If the screen saver has made the LCD display go dark, you will need to press one of the front panel keys to wake it up.

---

[1] While you can access Configuration screens from a Web browser (see page 21), values will not flash on the browser the way they do on ATM's LCD. For this reason, we recommend always configuring ATM from its front panel.

> **Don't walk away from Configuration Screens!** After five minutes of inactivity, the Configuration Screen times out and current changes are discarded. If you're tweaking the configuration and have to take care of something else, remember to press `DONE◁` before you leave.

Configuration screen menu map

`AUDIO◁`

Audio input, output, and level or sensitivity menus don't appear on Axia versions of ATM. On these systems, all audio is run over Livewire.

➢ `Audio input:` choose `Analog`, digital `AES/EBU`, or digital `s/pdif` (all input formats are stereo). Only one of these formats is available at a time.

- If you've selected `Analog`, you'll be prompted to adjust the `Input Sens.` in a later screen. This is important, to get the best signal/noise and distortion performance.

- If you've selected a digital format, input levels are inherent in the signal and you won't see the `Input Sens.` screen. If you must adjust these levels, do so in the device feeding ATM.

- Selecting AES/EBU or s/pdif changes both the data format and the electrical specification of the digital input connector. For more information, see page 39).

➢ `Digital out:` choose `AES/EBU` or `s/pdif`. This affects both the data format and the electrical specification of the digital output connector (page 36). The digital output is always active, even if you've chosen an analog input.

➢ `Input sensitivity` (shown only when Analog Input is selected). The LCD will look like this:

```
Input Sens.    NEXT◁
+14dBu=0dBFS   PREV◁
↑↓ Chg value
L=-10 R=-10    DONE◁
```

This screen lets you set the Analog input sensitivity. When you're done, press the softkey to go to the next screen.

Current value is +14 dBu: the absolute loudest analog signal ATM will see in your installation. Press the softkey to go to the previous screen.

Use the Up and Down Cursor buttons to change the current value

Reads the current Left and Right analog input, converted to digital, in dBFS. Press the softkey to save this setting.

- Select a value between `+20dBu` and `-10dBu` to result in fullscale within ATM and on the digital output.

- The bottom line of the display shows current Left and Right internal levels in dBFS. This is the result of the incoming analog signal, adjusted by the second

line of the LCD and sent to the analog-to-digital converter. Typically, these should read –12 for a +4 dBu lineup tone[1].

For most installations, where 0 VU = +4 dBu, we recommend setting the analog `Input sensitivity` **to** `+14dBu=0dBFS.`

Don't confuse dBu—an analog measurement—with digital dBFS!

Analog dBu is a comparison to an arbitrary voltage (.775 v), and signals can be much hotter. A console's output is typically +4 dBu (1.228 v) when the meter reads 0 VU, but that's for a sinewave test signal. Peaks in voice waveforms, in real-world broadcasting, can sometimes be considerably above that... even though properly-calibrated VU meters are too slow to register them. Analog is very forgiving of these momentary peaks.

However, in the digital world, 0 dBFS (decibels referred to Full Scale) is an absolute ceiling. Some devices distort horribly or crackle when an input attempts to pass this value. ATM handles overloads differently, applying serious limiting to gracefully protect your signal (and listeners).

If your facility uses the +4 dBu standard of modern broadcast equipment, our recommended setting of `+14dBu= 0dBFS` provides 10 dB of headroom for protection while still maintaining an 84 dB signal-to-noise ratio. With this setup, it's unlikely that ATM will ever go into distortion or audible limiting.

If you're dealing with heavily-processed feeds you can use a lower setting; for seriously dynamic performances it may need to be higher. ATM has 94 dB dynamic range at all settings, within 2 dB of the theoretical limit for 16-bit sound.

➢ `Analog Output Level`: shown when analog or digital inputs are selected. The LCD will look like this:

`Output Level   NEXT◁`	This screen lets you set the Analog input sensitivity. When you're done, press the softkey to go to the next screen.
`0dBFS=+14 dBu PREV◁`	Current value is +14 dBu analog output when the internal digital level is 0 dBFS. Press the softkey to go to the previous screen.
`↑↓ Chg value`	Use the Up and Down Cursor buttons to change the current value
`L=-10 R=-10    DONE◁`	Current Left and Right analog output in dBu. Press the softkey to save the setting.

- Select a value between `-10dBu` and `+20dBu` for the analog outputs, when ATM has 0 dBFS internally. This is usually set to match the Analog Input sensitivity, so our recommendation for most facilities would be `0dBFS= +14dBu`. However, you can use other settings if you want ATM to also create audio gain or loss.

- The analog outputs are always active, even if a digital input has been selected.

➢ `Default Comp`: This is the time compression ratio that first shows up on the Record and Play screens. Select between `0%` and the maximum selected in the next screen.

---

[1] The +14 dBu sensitivity in the figure would make these meters read –10 dBFS with a standard +4 dBu line level test signal. In other words, the setting means a +14 dBu signal would read 0 dBFS—the maximum allowable digital signal—and standard line level is 10 dB less than that, or –10 dBFS.

➢ `Max Comp`: This is the maximum compression that can be applied. For most flexibility, we recommend 20%. However, stations with some program formats might want to choose a smaller value. By reducing this value, you can insure that your operators never exceed the limit you set.

➢ `Ramp Time` (see the description of ramping under Play, page 16). Select between 1 second and 30 seconds. We recommend a ramp of 5 to 10 seconds for best results.

➢ `Hold Time`: This is the amount of time that will be added to `Holding:` for each tap of the Hold button. Hold time can be set from 5 to 300 seconds in 5 second increments. The `Holding:` time then counts down; when it reaches zero ATM returns to Play mode. You may also set this value to `Time Shift`, which suspends the countdown timer.

`NET‹`

> While broadcast engineers are becoming more and more net-savvy, you may need to consult with your station's IT manager about appropriate settings for your data network. ATM responds to "Pings" across your network, providing an easy way of determining if your DHCP or manual IP setup is working. Note that the act of enabling and disabling DHCP, then pressing DONE will refresh your network settings. It may take a few moments before a new IP address registers with ATM's network firmware.

➢ `DHCP:` Choose `enabled` or `disabled`. Enable if your network router is set up for DHCP address assignment. When DHCP is enabled, your unit's IP address, Subnet, DNS Server, and Gateway are supplied by the router.

NOTE: The following four screens appear only if you have DHCP disabled. Also, IP, Subnet and other server addresses "roll over" between their limits of 0 and 255: If the value `0` is showing, press the DOWN cursor to jump to `255`; if `255` is showing, press the UP cursor to jump to `0`.

➢ `IP Address`: When manually setting network addresses, use the LEFT and RIGHT cursor buttons to change fields; UP or DOWN cursor buttons to set values.

➢ `Subnet mask`: In most cases, this should be set to `255.255.255.0`.

➢ `DNS Server`: This setting will be used to allow ATM to check 25-Seven servers for updates and other future upgrades. For now, leave it set to 0.0.0.0

➢ `Gateway`: Supply the IP address of your LAN's Gateway system, so that ATM can connect to remote servers over the Internet.

➢ `NTP: Enables` or `disables` ATM's ability to set its time-of-day clock automatically from a Network Time Protocol server. If enabled, ATM's clock cannot be manually set. If you plan to use this feature, you should first manually set your clock and time zone to your local time before you enable NTP.

➢ `NTP server:` Enter the IP address of the network time server.

The following two menu items appear on Axia versions of ATM only:

➢ `AxiaSrcAddr:` Enter the Source address assigned to ATM by your administrator.

➢ `AxiaDstAddr:` Enter the Destination address assigned to ATM by your administrator.

About NTP

Network Time Protocol servers supply accurate time-of-day and other data over the Internet. ATM can use them to automatically set its internal clock. Your facility might also have its own NTP server.

NTP servers are often identified by a domain name—a familiar-sounding ".com" or ".gov" URL— rather than their numeric IP address. ATM requires numeric IP addresses.

You'll find a list of US servers, with both their domain names and their IP addresses, at the National Institute of Standards and Technology's website[1]. International time server domain names are listed at www.ntp.org. Please respect the access policies listed by the server provider and IT department. ATM users should never select a Primary server for NTP updating; use Pool or Secondary servers instead.

The best way we have found to test whether NTP services are working is to manually offset your ATM clock by a couple of minutes, then enable NTP. If your network connections are working and your NTP address is valid, the system clock should quickly acquire the correct time.

`GPIO◄`

➢ `GPIO Mode:` Select `Enabled` to activate General Purpose Input / Output control, a parallel remote control mode accessed through the DB-25 jack on the rear panel. Select `Disabled` if you have a remote connected to this jack but want to temporarily turn it off.

- GPIO wiring details are covered on page 36.

- ATM's GPIO system doesn't use noisy relays and won't be tripped by normal static impulses. You can leave it `Enabled` even if there is no remote control plugged in.

`TEST◄`

The GPIO test screen lets you verify that GPIO connections and external remotes are properly wired. It functions regardless of whether GPIO Mode is enabled or disabled.

---

[1] The list itself is currently at http://tf.nist.gov/tf-cgi/servers.cgi. If NIST has moved it from that location, use the search box on their home page, www.nist.gov, to find "Internet Time Service".

```
    12345678

Ins: 00000000 PREV◁

Outs:00000000

↑↓◁▷ Change    DONE◁
```

This identifies GPIO inputs and outputs in the two rows of zeros below.

These change from 0 to 1 while ATM is receiving a trigger on the DB-25 input (1-8) identified in the line above.

Use the Left and Right Cursor buttons to select a DB-25 output (1-8). Use Up and Down to change that output from 0 (off) to 1 (on).

This line reminds you to use the Cursor buttons to select an output and change its state.

When you leave the GPIO test screen, the DB-25 outputs are all reset to 0.

`INs◁ and OUTs◁`

The GPIO system has to be configured to determine which ATM functions appear at which pins on the DB-25 connector. To configure GPIO, press the `INs◁` or `OUTs◁` softkey. The LCD will look like this:

```
GPIO Inputs:

In1=Rec       PREV◁

↑↓ Chg value

◁▷ Chg field  DONE◁
```

This screen lets you map GPIO inputs to ATM functions. A similar screen lets you map GPIO outputs to ATM statuses and actions.

Sending a contact closure to input 1 activates Record mode. Press the softkey to return to the `GPIO Mode` screen.

Use the Up and Down Cursor buttons to change which function is assigned.

Use the Left or Right Cursor buttons to change the selected input. Press the softkey to save GPIO settings.

Each of the 8 Inputs or outputs (e.g., `In1`, `In2`, `Out1`, `Out2`, etc) can be assigned to an ATM function, as listed on the following chart. The same function can be assigned to multiple inputs or outputs, if you require isolated circuits.

GPIO FUNCTION	INPUT SETTING	OUTPUT SETTING
Nothing Assigned	None	None
Record Button*	Rec	Rec (pulse)§
Currently Recording		Rcrding
Play Button*	Play	Play (pulse)§
Currently Playing		Playing
Hold Button*†	Hold†	Hold (pulse)§
Currently Holding		Holding
Dropping to Real Time Play (standby)		RTP (pulse)§
Currently Real Time Play (standby)		InRTP
Dump all audio in buffer	Dump	Dump (pulse) §
Record Advance (dump and record)†	Advance†	
Delayed Record¶	DlyRec	
Delayed Play¶	DlyPlay	
Delayed Hold¶	DlyHold	

Delayed Dump¶	`DlyDump`	
Cue point #1¶	`Cue1`	`Cue1`
Cue point #2¶	`Cue2`	`Cue2`
... *(etc)*	*...(etc)*	*...(etc)*
Cue point #8¶	`Cue8`	`Cue8`
GPIO temporarily disabled because user is in a `CNFG⁕` screen		`Blocked`
Record Gate‡	`RecGate`	
Record Trigger‡	`RecTrig`	

* —Input mimics front panel button.

§—Output is pulsed 250 ms when function is first activated.

†—While you can press RECORD+PLAY on ATM's front panel to perform a Record Advance, this won't work from a remote because GPIO inputs are momentary. Instead, assign the dedicated Record Advance function (`Advance`) to a remote input.

‡—Record Gate (`RecGate`) and Record Trigger (`RecTrig`) work together to integrate ATM with satellite program services and your automation system. You can use them to automatically lengthen breaks and add availabilities. While RecGate is held low, pulsing RecTrig initiates record. If RecGate is left high or floating, RecTrig has no effect. If ATM is recording because of a RecTrig command, releasing RecGate puts ATM into Play mode. Normal front panel or remote buttons can override these states at any time.

For more on using these functions, see page 38.

¶—"Delayed" and "Cue" settings flag the audio as it comes into ATM. When the flag reaches ATM's output, the desired action takes place: either a transport activity takes place (such as Record or Dump), or a Cue GPIO output is triggered.

Since flags are locked to specific audio samples, they will always happen at the proper word cue and appropriate place in the program material, no matter how much time compression or delay has been applied during playback.

Flags can be set during all ATM modes, including Standby. If a flag is set during standby, the triggered action takes place almost immediately. (It takes about a quarter second for audio to pass from ATM's input to output in Standby, so the action is similarly delayed.)

Cue flags can be particularly handy when dealing with network programs. For example, if the network equipment provides a contact closure when it's time for a local break, connect it to an input that has `Cue1` assigned. If that closure is sensed during record[1], ATM will mark the current audio sample. During play-

---

[1] This includes sensing a closure in "Phantom Record" during Hold and Playback (page 16).

back, ATM will send a pulse to any output with Cue1 assigned when that sample is reached… where it can trigger your other equipment to play the break.

If you connect the network's "local break" contact closure to two inputs, one assigned to Cue1 and another assigned DlyHold, ATM will also enter Hold mode when the cue is reached during playback. This sets the compression ratio to 0% for the next 30 seconds, to assure that local breaks and network breaks are the same length even if the network program is compressed.

SYS‹

Pressing this softkey lets you adjust system settings

➢ Time Zone: Enter the time zone your facility is located in. If you plan to set the time of day manually, set the time zone first, then set the clock. If you are planning to use NTP (See Network settings) to automatically set your clock, first set the time zone, then manually set the clock. NTP will quickly adjust a clock that is just a few minutes off, but is slow to adjust clocks that are hours out of adjustment.

Time zone selections that include Daylight Savings Time (such as EST/EDT) will perform the necessary seasonal adjustments when connected to most network clocks. However, this won't happen if you're setting time manually (see next entry), since ATM won't know what day it is.

➢ Time of Day: Use the cursor buttons to set the time. Use the 12/24 soft key to set whether ATM's clock reads AM and PM or uses 24-hour international-standard notation. If NTP is enabled in the NET‹ settings, you will not be able to manually set the clock; Set by NTP will be displayed instead.

➢ Brightness: Adjusts the LCD backlight brightness.

➢ Contrast: Adjusts the LCD contrast for different viewing angles.

You might find it helpful to periodically re-adjust Brightness and Contrast, as the LCD and its backlight age.

➢ Screen Saver: Turns off the LCD backlight when in Standby mode, after a pre-determined time. If this value is set to 0 minutes, the LCD backlight never turns off; note that this may reduce the LCD's service life.

➢ Join Display: Sets the default view for whether the Play screen displays a minutes:seconds countdown to the join time, or displays the Rec@ time. Regardless of how this is set, you can switch freely between these views from the Play screen.

Info screens

When ATM is in standby mode and displaying Real-Time Play, you can press the INFO‹ soft key to view system settings.

Info screens generally report two settings each:

On XLR-equipped ATMs, you'll see

➢ `Input` [Analog, AES/EBU, or s/pdif] and `Digital Output` [AES/EBU or s/pdif].

➢ `Input Ref` (sensitivity) and analog `Out level` [both in dBu referenced to 0 dBFS].

On Axia-configured ATMs, you'll see

➢ `Axia Src Addr` Axia Source Address

➢ `Axia Dst Addr` Axia Destination Address

On all ATMs, you'll also see

➢ `Clock` time and `NTP` status. `NTP` will display `disabled`, or show the NTP IP server address if Network Time Protocol is enabled.

➢ `Network Address` and Subnet `Mask`

➢ `Serial number, Memory capacity` (as installed), and `Version` (software version and build date)

Installation

The Audio Time Manager is designed to fit in a standard 19" equipment rack, 2 units high by 12.5" deep (allow additional depth for the connectors). While it does not generate much heat, we recommend you allow at least one unit of blank panel for ventilation if a heat-generating device is mounted below it. Do not block the two vents on the rear of the unit.

The internal power supply will self-adapt to any voltage between 100 – 240 v 50/60 Hz. It consumes 1.8 A maximum; typical draw is about 32 watt @ 120VAC.

We recommend that ATM be connected to a UPS (uninterruptable power supply), to prevent loss of audio in case of a power disruption. This is good advice for any mission-critical electrical device.

Signal chain

ATM can be inserted anywhere in an analog or digital signal path where you want to control program timing and duration. But for maximum flexibility and efficiency, we recommend using a routing switcher: have every source you'd want to time-control (such as remotes and network feeds)connected to an input, and at least one output for ATM and one for monitoring. Connect ATM's output directly to a console input channel, or back through another router input. Obviously, you need stereo connections at the router if you intend to use ATM for stereo signals.
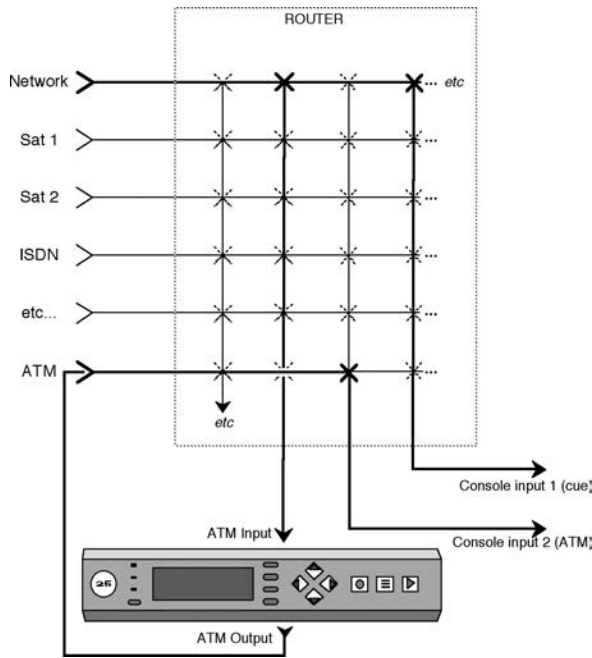
In many cases, this router already exists in your facility. All you'll need to do is provide the mono or stereo input and output for ATM.

In the drawing, the network feed is going to both ATM and console channel 1. Channel 1 feeds just a cue speaker, so the operator can hear the network while ATM is recording.

Typical router installation. Active circuits are shown with bold lines. In this case, we're managing the timing of a network program, so the network is fed to both ATM's input and also to a console channel for operator cueing. ATM's output is fed to a different console channel and then to air. (If you want to use one less router input, you can send ATM's output directly to a console input.)

Rear Panel
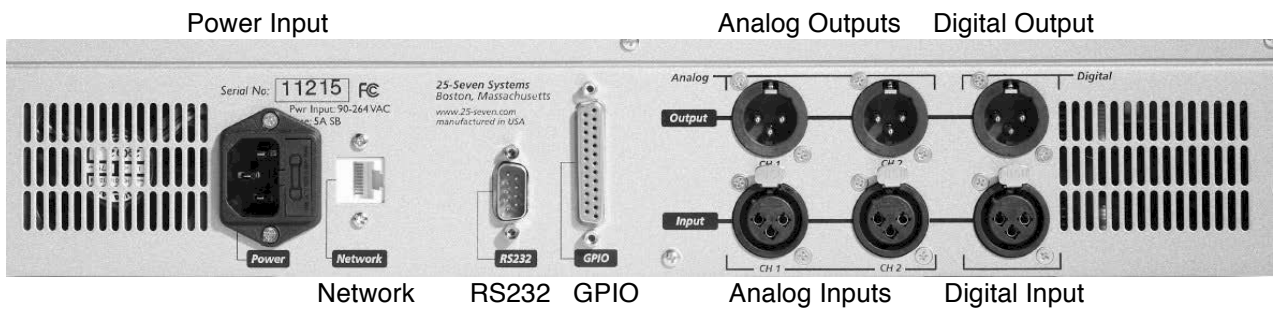
Power Input

Standard IEC configuration self-adapts to any voltage between 100 – 240 volts AC, 50 or 60 Hz.

Network

Connect to a standard 10/100Base-T network connection. This port is used for software updates, synchronization to a network time server, and secure remote control via a web browser. If connected to the Internet, it should be behind a hardware firewall.

On Axia-configured ATMs, this port is also used for Livewire audio over IP.

Analog Outputs

These are electronically balanced XLR male, pin 2 hot, designed to feed a load of 600Ω or greater. For information on levels see page 27.

Unbalanced operation is not recommended.

ATM uses advanced algorithms to process signals in true stereo, with no smearing or variation in the stereo field during time compression. So while the analog outputs (and analog inputs) are labeled CH1 and CH2 on the back panel, we assume you'll be using them for the left and right channels of a stereo pair and that's how we've written this documentation.

However, you can use ATM for dual, identical processing of two mono signals. There's no cross-talk.

Digital Output

When set to AES/EBU via the configuration menu, this output conforms to IEC 958 Professional (5v p-p, 110Ω balanced) on XLR connectors. Output is 16-bit PCM linear.

When set to s/pdif, the voltage and impedance switches to IEC 958 Consumer (.5v p-p, 75Ω unbalanced): connect signal to pin 2 and shield to pin 3 only. Do not connect pin 1 for s/pdif operation.

> This digital output is always active, so you can use ATM as the converter if you have to handle an analog signal in a digital plant.
>
> If there is no signal on the digital input jack, ATM provides its own low-jitter clock at a 44.1 kHz sample rate. When there is a valid 32 kHz, 44.1 kHz, or 48 kHz signal at the digital input jack—even if analog inputs are selected—the internal clock and digital output are locked to it. Use this feature to make sure ATM's output works smoothly with your routers and other equipment. But see the warning about digital clocking on page 39.

RS-232

This jack is used for serial remote control, and configured as a computer DTE port similar to the Comm 1 jack on most PCs. ATM communicates at 9600 baud, no parity, 8 bits, 1 stop bit, no flow control (usually referred to as "9600N81".

If you want to control it from a computer or dumb terminal, you'll need a null modem or a cable that reverses pins 2 and 3, pins 4 and 6, and pins 7 and 8.

Details of the command language are on page 40.

GPIO

Eight parallel control inputs and eight parallel control outputs appear on a DB-25 connector, with pinouts detailed below. Input and output functions are assigned through a CNFG4 menu; details are on page 30.

Inputs and outputs are opto-isolated for easy interface to other equipment. A +5v supply and ground are also brought out to the DB-25 for simple remote controls using pushbuttons and LED status readouts.

The +5v supply can carry 200 mA, more than adequate for 8 LEDs and 8 logic inputs. It is protected by an internal, self-resetting thermal circuit breaker.

Basic remote control:

1)   Use the connections shown in gray on the left side of the figure that follows. At the DB-25, jump pin 1 (input common) to pin 6 (+5v).

2)   Connect one side of an SPST NO pushbutton to pin 19 (ground); connect the other side to pin 14 (Input 1).

3)   Add up to 7 more pushbuttons as needed, connecting one side of all to pin 19 as common, and connecting their other sides to the individual inputs in the chart below.
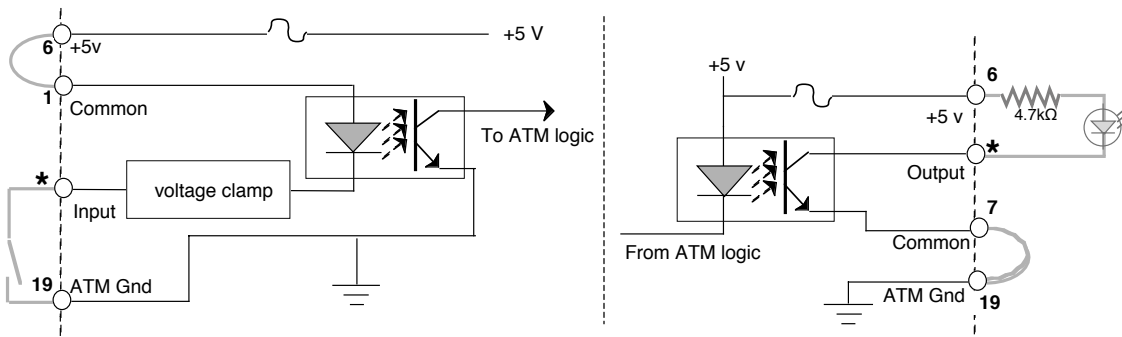
ATM typical remote input (left) and output (right) circuits. Use the connections shown in gray for simple pushbutton control. Or use just the opto-isolator connections without pins 6 and 19 for interface to other equipment. DB-25 pinouts shown in bold. * indicates individual DB-25 pins for the 8 inputs and 8 outputs; see chart below.

Basic remote readout:

1)   Use the connections shown in gray on the right side of the figure. At the DB-25, connect pin 7 (output common) to pin 19 (ATM ground).

2)   Connect a 4.7 kΩ 1/4 watt resistor between pin 6 (+5 V) and the anode or + lead of an LED; connect the cathode or – lead of the LED to pin 20 (output 1).

3)   Add up to 7 more resistor-and-LED combinations as needed, using pin 6 as the common voltage supply and connecting the LEDs cathodes to individual outputs in the chart below.

**Connecting to external logic or relay circuits:**

ATM's GPIO inputs and outputs are isolated from its power supply and ground, and can be connected to external logic circuits or the power supplies in remote equipment:

• Inputs require at least 6 mA flowing from the input common (pin 1) and the individual input circuit pin.

• Outputs can carry up to 100 mA, up to 30 V DC, between the individual output circuit pin and output common (pin 7).

• Outputs may conduct for a few seconds during ATM power-up, as part of a self-test routine. Be aware of this if you are restarting ATM while the GPIO is connected to other equipment.

GPIO Pinout

PIN	SIGNAL	PIN	SIGNAL
1	Input Common	14	Input 1
2	Input 2	15	Input 3
3	Input 4	16	Input 5
4	Input 6	17	Input 7
5	Input 8	18	NC
6	+5 VDC	19	Ground
7	Output Common	20	Output 1
8	Output 2	21	Output 3
9	Output 4	22	Output 5
10	Output 6	23	Output 7
11	Output 8	24	NC
12	Ground	25	NC
13	NC	NC — No internal connection	

Automatic operation with Satellite Receivers
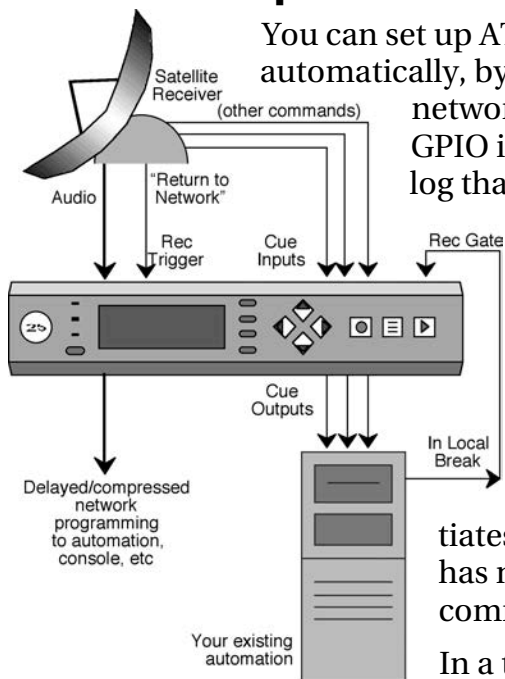
You can set up ATM to lengthen local breaks in network programming automatically, by wiring contact closures from your satellite or network cue receiver and automation system to ATM's GPIO inputs. More spots can be added to the automation's log than would normally fit in the break, and ATM will delay the return to network exactly long enough to accommodate them, without losing any network content, and without operator intervention.

The key to making this work is using ATM's `GPIO` configuration screen to assign Record Gate (`RecGate`) and Record Trigger (`RecTrig`) commands to those GPIO inputs, as discussed on page 32. While RecGate is held low, pulsing RecTrig initiates record. If RecGate is left high or floating, RecTrig has no effect. If ATM is recording because of a RecTrig command, releasing RecGate puts ATM into Play mode.

In a typical application, connect the GPIO input assigned to RecGate to an output relay of your automation system. Configure the automation system to close that relay when a local break starts, and release it when the break is over. Your automation manufacturer can help you develop the proper commands, which may need to be manually inserted in the log.

Connect the satellite's "Return to Network" relay closure to the ATM GPIO input that's been assigned Record Trigger. Route the satellite's audio through ATM, and then to your console or the appropriate input on your automation system.

Any additional network cue relay closures that would normally go to the automation system can be routed through ATM. Wire the network cue output relays to GPIO inputs, assigned to numbered Cue flags (page 30); wire GPIO outputs with matching Cue flag assignments to the automation inputs. By running these relay closures through ATM, network cues are kept in perfect sync against the audio, even when ATM is delaying or compressing the programming.

Here's how this setup can work:

- When the network sends a "Begin Local Break" command, your automation system plays the log entry to close the RecGate relay, and begins playing local spots.

- The automation system keeps the Record Gate relay closed as long as local spots are playing, even if there are more spots than the network break would normally accommodate. When the network sends "Return to Network", it activates Record Trigger and ATM starts recording network programming.

- When the local break is over, your automation system sends the command to release the Record Gate relay. This puts ATM into play mode, seamlessly joining the network and sounding as if they had given you a longer break.

Analog Inputs

These are electronically balanced XLR female, pin 2 hot, with a load of 20kΩ: this makes it compatible with all modern electronically-balanced outputs. If fed from a transformer-balanced output, we recommend bridging a 680Ω resistor between pins 2 and 3. For information on levels see page 27.

Unbalanced operation is not recommended.

Digital Input

When set to AES/EBU via the configuration menu, this input conforms to IEC 958 Professional (5v p-p, 110Ω balanced) on XLR connectors.

When set to s/pdif, the voltage and impedance switches to IEC 958 Consumer (.5v p-p, 75Ω unbalanced): connect signal to pin 2 and shield to pins 1 and 3.

> **Warning:** ATM locks to any valid 32 kHz, 44.1 kHz, or 48 kHz signal at this connector, even if you've selected analog for the audio input. This lets you keep ATM's digital output compatible with other equipment. But because of this feature, both the analog and digital outputs can be disrupted if ATM sees an intermittent or weak digital input signal. **Do not connect, remove, or change a digital input signal while ATM is on the air.**

Under extreme circumstances, an intermittent digital input can cause erratic audio behavior. If this happens, power-cycle the ATM.

Serial Control Language

ATM can be controlled by a computer connected to the DB-9 jack on the rear panel; wiring and data format details are on page 36.

We deliberately designed ATM's control language to be simple: all that's needed are single keystrokes, or trivial routines in any computer language.

Commands are sent as single ASCII characters, case-insensitive, with no carriage return or linefeed: for example, if you've connected ATM to a terminal program, all you have to do is hit "R" to start recording.

Commands are executed immediately upon receipt, and are not echoed.

Transport Commands		Compression Settings	
cmd	action	cmd	setting
A	Record Advance	0	None (Delay)
R	Record	1	2 mins/hr (3.3%)
P	Play	2	3 mins/hr (5%)
H	Hold	3	4 mins/hr (6.6%)
D	Dump	4	5 mins/hr (8.3%)
S	Delayed Record	5	6 mins/hr (10%)
T	Delayed Play	6	8 mins/hr (13.3%)
I	Delayed Hold	7	10 mins/hr (16.6%)
E	Delayed Dump	8	12 mins/hr (20%)
U	Set Cue 1		
V	Set Cue 2		
W	Set Cue 3		
X	Set Cue 4		
Y	Set Cue 5		

See explanation of "Delayed" settings on page 32.

Axia Configuration

Introduction

ATM's Version 2.2 Axia Software gives you the benefits of Audio Over IP, with complete integration in an Axia / Livewire environment. Basic operation requires just an Ethernet connection to an Axia system. High-quality audio gets carried on that same network cable and standard computer components route it to the right destination in your facility. No additional audio wiring is needed; in fact, Axia-equipped ATMs don't even have active analog or AES/EBU audio ports.

All controls, and most of the menus, are identical to the standard Audio Time Manager. Operators won't have to learn anything new. Installation and setup is almost the same, with a few important exceptions. We recommend you read the main ATM manual first. Once you're familiar with ATM's functions and menus, use this supplement to understand Axia-specific setup.

Network Setup

Since ATM-Axia uses network connections instead of standard analog or digital audio wiring, it won't operate until it's been properly configured for your network. An IP address must be assigned, and Livewire source and destination channels have to be specified.

Addresses

Using the softkeys (page 29), press CNFG◁ and then NET◁ to open the Network menu. Set up DHCP, IP Address, and Subnet Mask according to the instructions on page 29. You may need to contact your network administrator for specific information about your installation.

Once you've done this setup, use the INFO◁ softkey (page 34) to verify ATM's IP address. Then you can check the connection by Pinging that address from another computer on your network.

If you are using DHCP and the INFO screen says the IP Address is 0.0.0.0, try assigning an unused static address that is valid for your network. Power cycle ATM after making and saving this change.

Axia Channels

Since ATM-Livewire uses network connections instead of standard analog or digital audio wiring, it won't operate until it's been properly configured for your network.

ATM's output audio is broadcast to the network as a single **Livewire Source Channel,** identified by a number between 1 and 32,767. This number should be unique on your network, and not be assigned to any other source channel.

Go to the Network menu (page 29) and select `Axia Source`. The LCD will look like this:

```
Axia Source    NEXT◁

05001          PREV◁

↑↓ Chg value

◁▷ +/- 100     DONE◁
```

This screen lets you set the channel number for ATM's output as a Source. When you're done, press the softkey to see the next screen.

Current value is 05001. Other devices on the network can listen to ATM at this address. Press the softkey if you want the previous screen.

Use the Up and Down Cursor buttons to change the Source value.

Press the Left and Right Cursor buttons to increment the Source value by 100 at a time. Press the softkey to save this setting.

ATM's input is fed by specifying a **Livewire Destination Channel** number. This is the Source Channel number of some other device on the network. It's often the output of a console, but it can be a network or remote, or any other signal source.

Select `Axia Dest` from the network menu. The LCD will look like this:

```
Axia Dest.     NEXT◁

01003          PREV◁

↑↓ Chg value

◁▷ +/- 100     DONE◁
```

This screen lets you set which signal goes to ATM's input.

This must be set to Source address of some other device on your Axia network. In this case, it's 01003.

Use the Up and Down Cursor buttons to change the input address.

Press the Left and Right Cursor buttons to increment the value by 100 at a time. Press the softkey to save this setting.

Remember: on the `Axia Dest` screen, ATM is the destination. Enter the channel number of some other source you want ATM to process.

Once you've set an address in this screen, ATM checks the network to make sure it's a valid source. If it is, the green front panel AUDIO LED will light.

- The LED shows that ATM is connected to a source on the network and is ready for use. It doesn't necessarily indicate that there's program audio on that source—it may be silence, while waiting for a network feed or a remote to start—but that a good connection has been made.

When both Source and Destination channel numbers are correct, ATM is ready for Livewire audio. You can check the signal flow by putting ATM in real-time play, feeding it a signal, and listening to ATM's Source Address on any Axia control surface or other monitor point.

If you are controlling your network with **Pathfinder**™ routing and control software, you have to register ATM in Pathfinder's configuration controls. If Pathfinder doesn't know ATM is on the network, you won't be able to use that software to route its audio.

Troubleshooting

During startup, ATM's LCD will look like this for a moment:

```
Waiting for

Audio stream
```
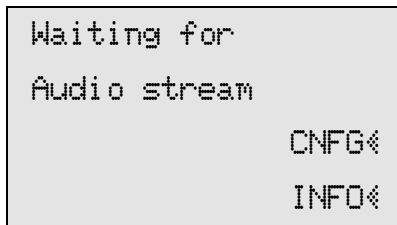
This means that ATM is logging on to the Livewire network.

`CNFG◄` Use this softkey to access the Configuration menu.

`INFO◄` Use this softkey to check ATM's setup info.

The screen will go away after ATM has successfully logged onto the Livewire network. If it remains on for several minutes after powering up, check the following:

- Does the `INFO` screen show a valid IP address?

    - If not, and you're using DHCP, try entering a manual address and restarting

- If the address is valid, can you Ping this address from another machine on the same network segment?

    - If not, check network connections, cables, and switches.

- Is ATM as a destination (`Axia Dest`) set a valid *source channel number* for some other device on your Livewire network? If so, the Audio LED will light green, and the `Waiting for Audio Stream` message will disappear. Note that silence is just as valid a signal as tone or music, so don't mistake lack of program material for a lack of connectivity.

Once you're sure audio is being sent to ATM, the unit will start in Real-Time Play mode. Now you should be able to monitor its Source address (as set in the `Axia Source` screen) from other devices on the Axia network. Verify meter and monitor levels to and from the ATM at your Livewire node or control surface.

Avoiding Trouble

ATM's link to the network should never be disconnected while the unit is in normal operation. Loss of network connection while the unit is operating will result in a loss of audio, and `Waiting for Audio stream` gets displayed on the LCD.

Once the network connection is restored, ATM will come up in Real-Time Play.

If the network connection was only out momentarily, but there was a lot of audio already in ATM's memory, you may be able to recover audio that was sent to ATM before the interruption.

1)   Press the RECORD button.

2)   Press the `CUE◄` softkey.

3)   Rewind back in time to before the outage took place (see page 13).

4)   Then press PLAY.

Specifications

Audio:

> S/N = 84 dB with 10 dB headroom (94 dB dynamic range), A-weighted
> THD @ 1 kHz = .01%; IMD (IHF) = .01%
> Frequency response = ±.5 dB 20 Hz – 20 kHz
> Measured with analog i/o; expect slightly better results with digital i/o.
> No change in audio quality with increased time compression.

Dimensions: 19"x12.5"x3.5".

Weight: 7.5 lbs (chassis).

Power: 100 –240 v 50/60Hz, self-adjusting. 1.8 A maximum draw; typically 32 watt @ 120VAC.

For more Information

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Warranty

25-Seven Systems, A Telos Alliance Company (hereinafter "25-Seven" and "we" or "us") warrants Audio Time Manager ("ATM" and "the unit") to be free from defects in material and workmanship for a period of one year from the date of receipt by the end-user.

This warranty is void if the unit is subject to Acts of God including (without limitation) lightning; improper installation or misuse including (without limitation) power line surges; accident; neglect or damage.

EXCEPT FOR THE ABOVE-STATED WARRANTY, 25-SEVEN MAKES NO WARRANTIES, EXPRESS OR IMPLIED (INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE).

In no event will 25-Seven, its employees, agents or authorized dealers be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of ATM or the inability to use ATM either separately or in combination with other equipment or materials, or from any other cause.

In order to invoke this Warranty, notice of a warranty claim must be received by 25-Seven within the above-stated warranty period and warranty coverage must be authorized by us. If we authorize the performance of warranty service, we will supply a shipping address and return authorization number: the defective unit must be delivered to that address, shipping prepaid and with that authorization number clearly marked.

We at our option will either repair or replace the unit and such action shall be the full extent of 25-Seven's obligation under this Warranty. After the unit is repaired or replaced, we will return it to the party that sent the unit and will pay the cost of ground shipping.

25-Seven's authorized dealers are not authorized to assume for us any additional obligations or liabilities in connection with the dealers' sale of the unit.