Smart pi ECOsystem



Smart Public Information for the smart city.

Smart pi takes information from across the city and delivers it to the people who need it: as an announcement, on a display, through a help point or straight to a hearing aid, the moment it matters.

One ecosystem, designed to look, feel and work as one

Every product works on its own, and better together.

IN

Where information comes from

- RTA
- MIC
- PBX
- SPL
- Fire panel

SMART PI CORE

Decides what to say, where and when

- DVA
- DVA-GTFS
- EYE
- DSP
- SIP
- FSM
- LOG
- NMS

OUT

How it reaches people: hear, see, ask

- NAM
- NAM-HP
- EDC
- EYE-HD
- HLP

Smart pi

SIP audio bridge

Connect sound card channels to SIP calls, across the building or across the network.

Crystal-clear audio. A connected ecosystem.

SIP connects the endpoints. OPUS carries the sound. Smart pi brings the audio paths together.

SIP (Session Initiation Protocol) establishes, manages and ends calls between networked endpoints. Smart pi SIP Server bridges those calls to sound card channels, connecting local Dante or AES67 audio with remote sites through physical or virtual audio devices.

OPUS encodes the audio for crystal-clear speech and high-quality music, with efficient bandwidth use. Adaptive receive buffering helps manage changing network conditions. Together, they extend paging, announcements and help point audio across routed networks and internet links.

MIC + DSP

Connect configured paging and zone audio paths.

NAM

Reach remote listening areas through the NAM SIP option.

HLP + DVA GTFS

Link help point calls with service announcements.

PBX

Use telephony call routing, or connect peers directly.



Many channels, one view
Native radial metering for sound card inputs.

512
Up to 512 audio channels, subject to device and licence.

48 kHz
Audio processing with 24-bit sound card support.

OPUS
Clear speech and music; G.711 for compatibility.

2-way
Sound card audio to and from SIP calls.

Native SmartPi UI with fictional endpoints and illustrative call, audio and network data.

From the sound card to the remote audio path

Keep channel routing, call control and network transport understandable.

Source
Sound card input

→

Extension
Channel mapping

→

SIP link
PBX or direct

→

Remote peer
Compatible endpoint

→

Destination
Sound card output

- Each extension has its own capture and playback channel mapping. The return audio follows the configured reverse path.

Registered through a PBX

Use the Smart pi PBX or another standard SIP server for registration and the configured call-routing arrangement.

Point-to-point

Connect compatible endpoints directly when the installation does not need a PBX in the call path.

Give each product its role

Smart pi SIP Server handles the bridge between audio devices and SIP calls. Smart pi PBX provides the telephony functions, including extension routing and trunks.

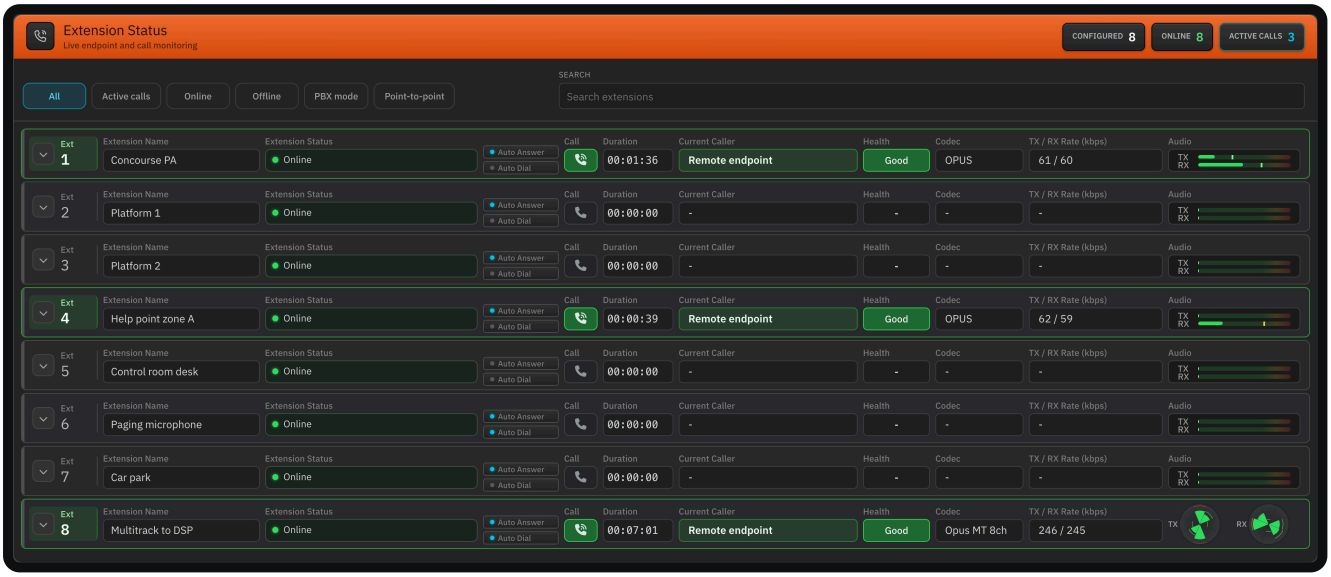
Start with the path

- Identify the source input and destination output.
- Choose registration or point-to-point operation.
- Agree codecs, call direction and remote endpoint compatibility.
- Test both audio directions on the actual network.

ⓘ Illustrative configured audio path, not a wiring drawing.

See the calls. Follow the audio.

Bring endpoint availability and media activity into one operating view.



Extension status
Active and idle calls, audio activity and a multitrack link in the same view.

Call state at a glance

Review online status, call duration, codec, TX / RX rate and audio levels. Filter the list to active calls or search for an endpoint.

Activity in both directions

Transmit and receive meters help separate a connected call from a working audio path. An idle extension remains visible alongside active calls.

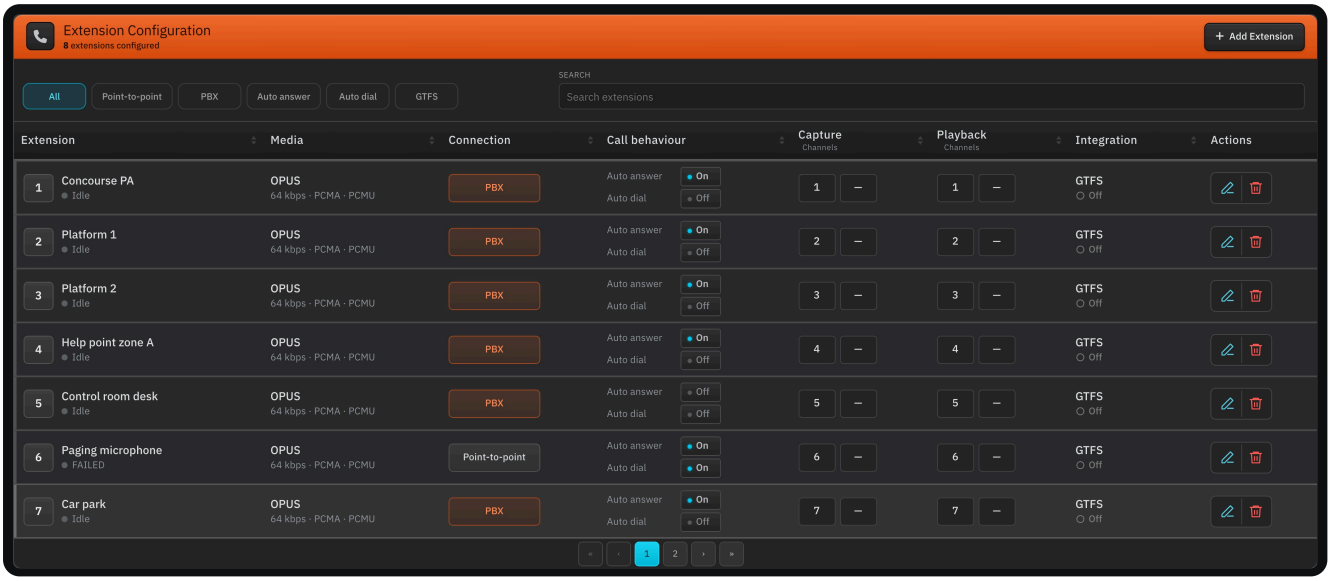
Investigate the right stage

A registered extension, an established call and useful audio are separate checks. Review each stage, then expand the call for timing, recovery and audio details.

Native SmartPi UI with fictional endpoints and illustrative call, audio and network data.

The right source.
The right destination.

Make each extension’s audio mapping and connection behaviour explicit.



Extension configuration
Media, connection mode, call behaviour and channel routing together.

Route deliberately

Assign each extension to the required capture and playback channels. Verify direction, mono / stereo selection and channel ownership.

Set the call behaviour

Choose auto answer, auto dial with automatic reconnect, and speed dials for the intended operating procedure.

Choose the media for the peer

Set codec priority and OPUS parameters per extension, including bit rate and mono, stereo or dual-mono operation. G.711 A-law and µ-law support compatible telephony endpoints.

Channel count is an installation choice

The published maximum is 512 audio channels per server. Actual capacity depends on the sound card, licence, media settings and workload; validate the intended system.

Native SmartPi UI with fictional endpoints and illustrative call, audio and network data.

Many channels. One clear overview.

Find activity across the sound card, then inspect the individual audio path.



Sound card inputs
A 64-channel sample bank with active input levels.

Scan the bank

Radial metering brings many channels into one view. Select the channel you need, then compare its assignment with the extension and call state.

Follow the direction

TX represents sound card inputs sent into calls. RX represents received audio routed to sound card outputs. The monitoring view separates these paths.

Listen with context

Select a source for monitoring, review its call health and confirm that the audio belongs to the intended route.

Physical or virtual audio

Use ALSA on Linux, ASIO or WDM on Windows, or a suitable virtual Dante / AES67 sound card.

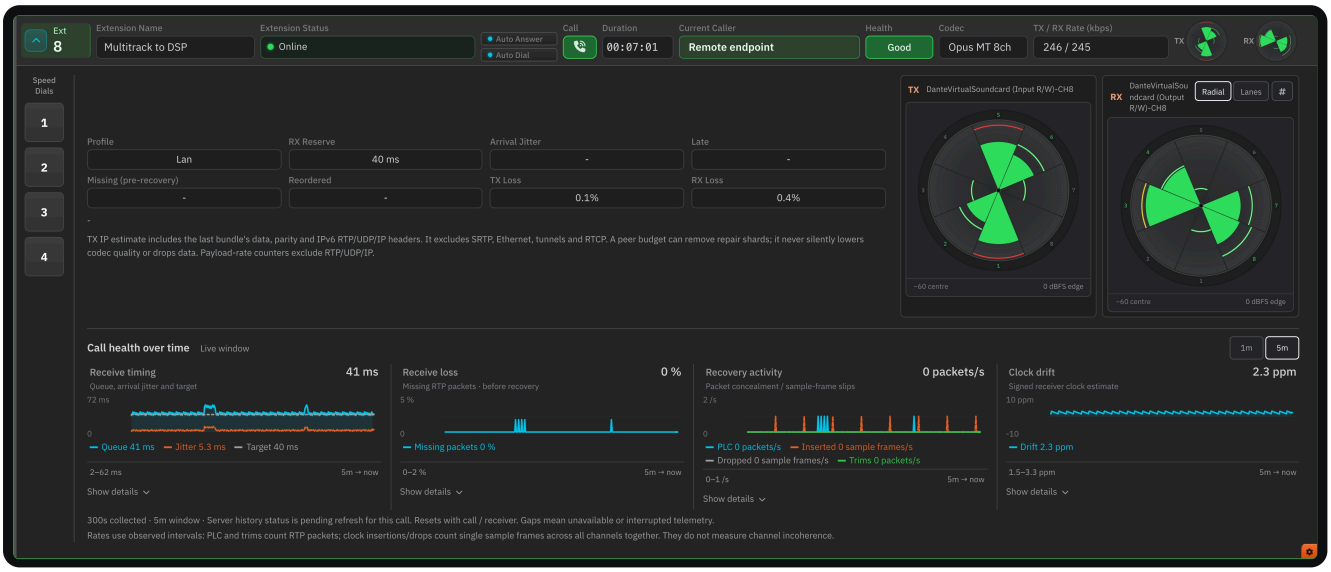
Check the whole route

Meters establish activity at a point in the system. Confirm the remote output, amplifier and listening area during commissioning.

Native SmartPi UI with fictional endpoints and illustrative call, audio and network data. Browser simulator audio is generated demonstration material.

Several channels. A coordinated audio link.

Use the multitrack view to inspect the channels and the work needed to carry them.



Expanded multitrack call
An eight-channel sample link with individual TX / RX audio and call details.

Keep the channels separate

Private OPUS multitrack trunks carry several channels in one compatible link. Review each track's activity and the mapped sound card channels.

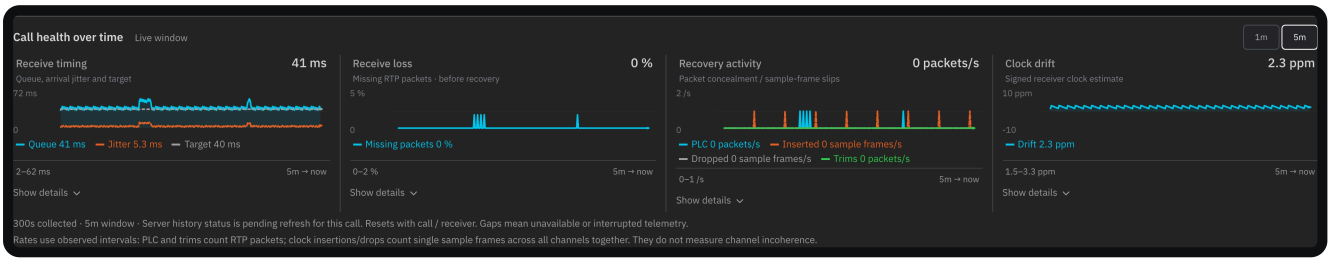
Watch the encode budget

The expanded call exposes decoder / encoder workload and protection information alongside the audio. Use these when validating the chosen channel count.

Native SmartPi UI with fictional endpoints and illustrative call, audio and network data. Private multitrack requires compatible peers; it is distinct from standard SIP mono / stereo interoperability.

Adapt to the link. See what changed.

Use time-based evidence to assess an audio path, rather than relying on one current value.



Call health over time
Compare the timing and recovery trends for the active sample call.

Profiles for the link

Choose Auto, LAN, WAN, 5G / mobile or 4G / resilient. All five are adaptive receive profiles, with different scheduling reserve ranges.

Recovery with evidence

Available OPUS FEC and private multitrack protection depend on negotiation and media conditions. Review recovery counters with packet loss and call health.

Reserve is not end-to-end latency

RX reserve is receiver scheduling headroom. Decoder and device queues, network transit and the remote system also affect the time to the loudspeaker.

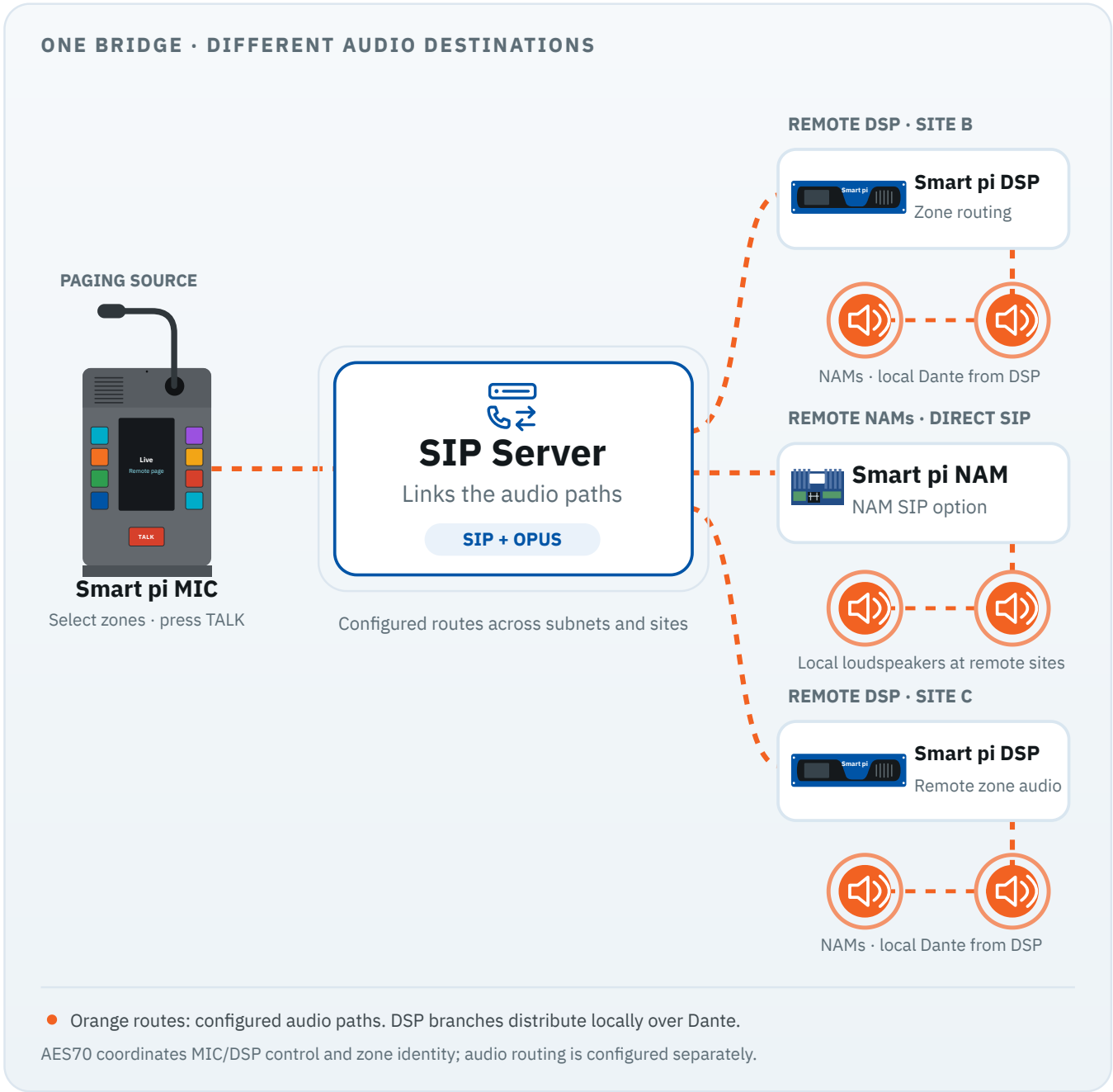
Compare saved and active settings

Some receive-profile and recovery changes take effect on the next call. Confirm the active call settings after reconnecting and test the link under representative network conditions.

Native SmartPi UI with fictional endpoints and illustrative call, audio and network data. The production server retains up to five minutes of active-call health in memory.

One SIP Server. Connected listening areas.

Link MIC paging to remote DSP zones and NAMs fitted with the SIP option, through one audio bridge.

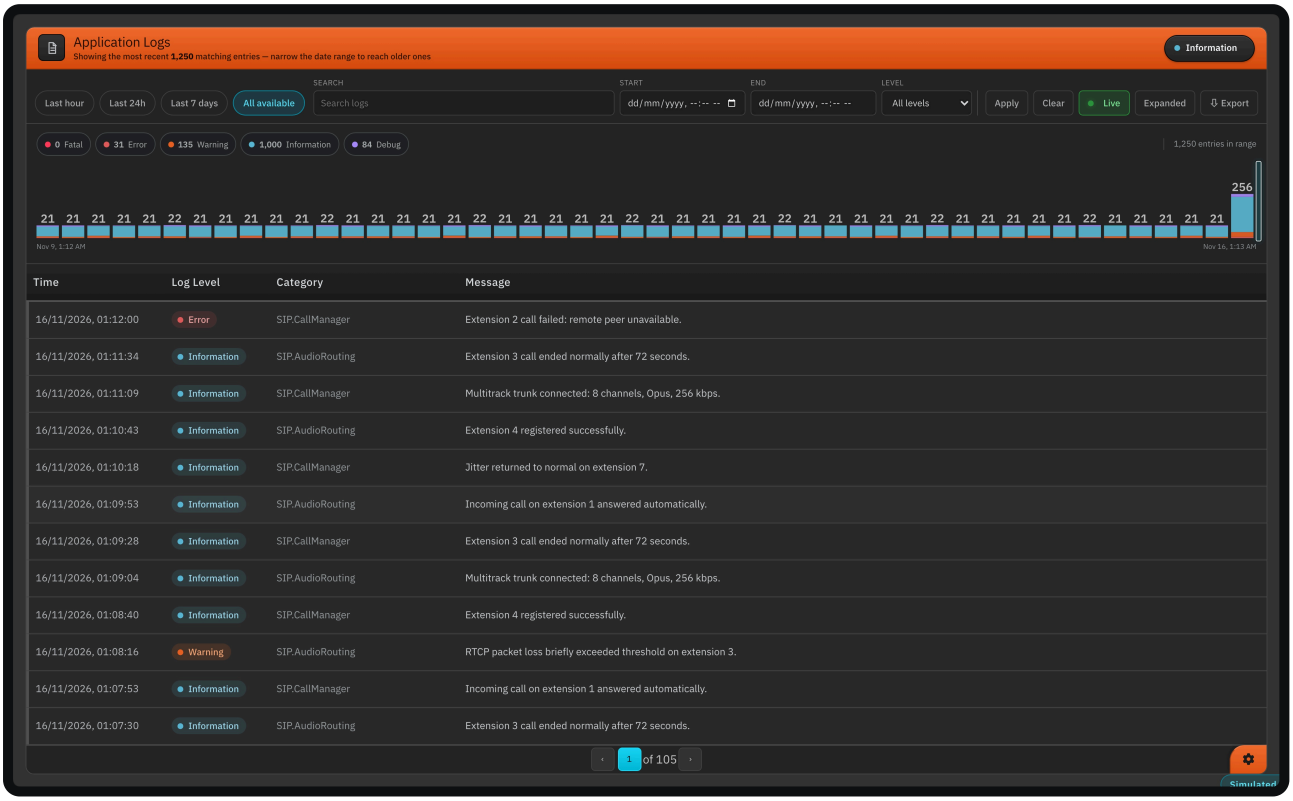


Configured routes to DSP zones and remote NAMs.
Via DSP, MIC connects before the chime and live speech. Direct NAM paths use their configured SIP routing.

Illustrative configured installation. Compatible MIC/DSP integration, NAM SIP options, SIP extensions, audio routing and network reachability are required.

From a symptom to the relevant events

Use event history with the live call and audio views to narrow an investigation.



Events over time
Filter by severity and period, then inspect the entries behind the timeline.

Find the interval

Use the timeline to locate changes in event volume, then narrow the date range, severity and search text.

Keep access accountable

LDAP / Active Directory group permissions and API keys support controlled access. REST API and AES70 provide remote control and monitoring.

Match the event to the symptom

Compare the log interval with endpoint availability, call health and channel activity. Configure retention around the operational review process.

Native SmartPi UI with fictional endpoints and illustrative call, audio and network data.

Map it.
Connect it. Prove the audio.

Build the setup around the required audio paths and the network that carries them.

1. **Map the audio**
   List sound cards, source inputs, destination outputs and the required mono / stereo or multitrack arrangement.

2. **Define the calls**
   Choose PBX registration or point-to-point operation, endpoints, codecs and automatic call behaviour.

3. **Set the network policy**
   Choose receive profiles for the actual link, then test loss, jitter, reconnect and recovery behaviour.

4. **Secure and integrate**
   Configure permitted users, LDAP groups and API access. Validate any AES70 or application integration.

5. **Accept the full path**
   Test both audio directions, channel identity and remote output, then retain the accepted configuration.

Windows or Linux

Runs on .NET with physical or virtualised deployment, on premises or in the cloud. Select the supported audio driver for the platform.

Size for the media workload

Channel count, OPUS settings, multitrack protection and audio drivers affect the workload. Validate the target system rather than treating the channel ceiling as a benchmark.

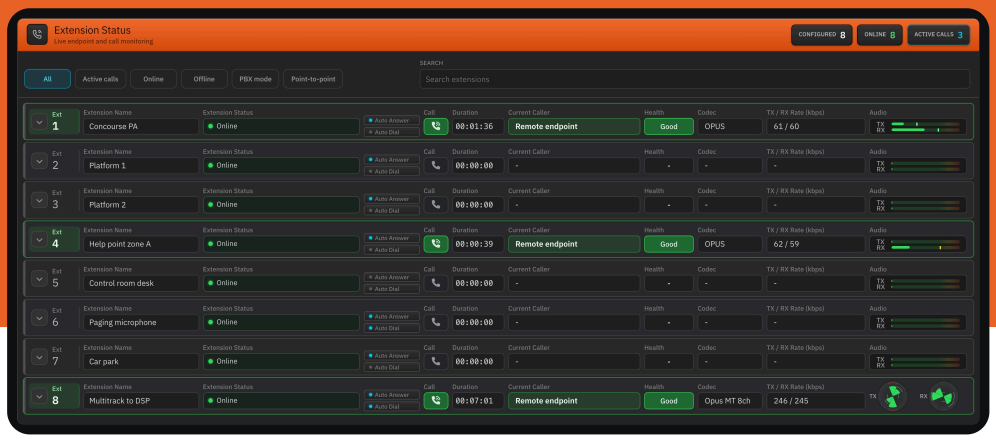
Include a recovery test

Interrupt and restore the intended network path. Check call reconnect, active settings, log entries and useful audio at the destination.

SMART PI SIP

Connect the audio. Reach the destination.

Talk to our team about your audio channels, remote sites and SIP integration.



Web
www.smart-pi.info

Email
info@smart-pi.info

Sales in Australia and New Zealand: Qvest
+61 2 8011 0500

Scan to visit smart-pi.info

Smart pi ECOsystem

Smart pi is a T&M Media Pty Ltd brand · ABN 67 612 111 369
Native interface with fictional sample calls. Specifications may change without notice.