Telematics IP-TCX — Feature List

Hybrid analog + IP telephone exchange

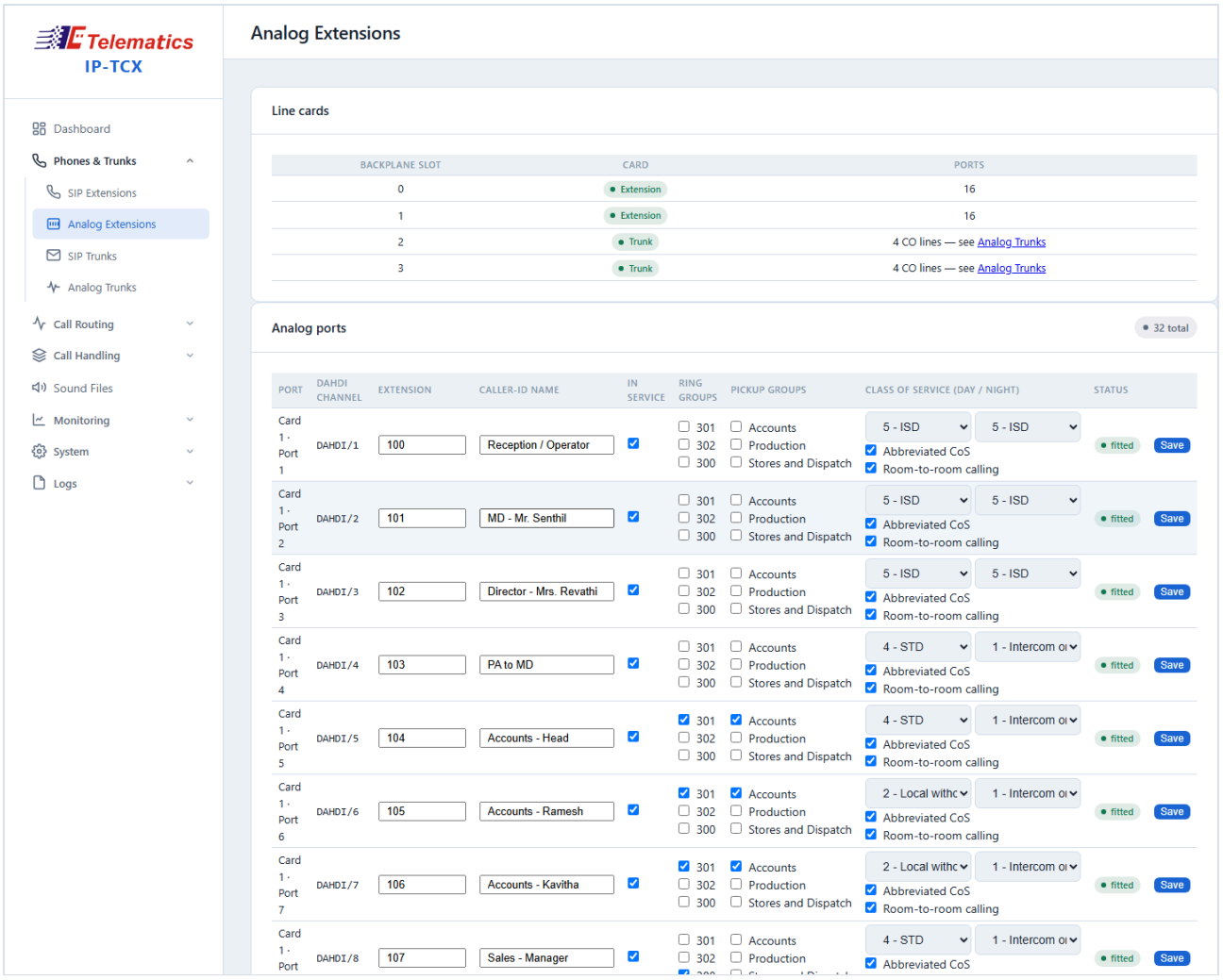
The IP-TCX is a hybrid telephone exchange. It keeps everything an office already relies on, which is analog desk phones on ordinary extension wiring and analog CO lines from the exchange, and adds SIP phones, mobile softphones and SIP trunks from BSNL, Airtel and other providers. One web page in a browser sets up the whole system.

*It works the way people who know a traditional EPABX expect. Phones use the familiar codes (0 for a line, 8N for a particular line, flash to transfer, *8 to pick up). There is a class of service ladder from "not connected" to "ISD", with separate day and night levels, plus Global Memory numbers, night service on each line, and pickup groups of up to eight. Nobody has to learn dialplan syntax: numbering is chosen from lists, and the system refuses any setting that would clash with a number already in use.*

This list describes the software in service from November 2026.

1. System at a glance

Item	What the IP-TCX provides
Analog extensions	Line cards of 16 ports each, up to 8 cards (128 analog phones). Ports are detected from the cards actually fitted.
Analog CO lines	Trunk cards of 4 lines each, up to 2 cards (8 CO lines). Incoming caller ID is decoded on the card.
SIP extensions	Desk SIP phones, softphones and mobile apps, alongside the analog ports, on the same numbering plan.
SIP trunks	Registration (username and password) or IP-authenticated trunks, with a backup trunk per route.
Local analog calls	Analog-to-analog calls are connected in the line-card hardware and use no voice channel to the processor.
Analog-to-IP calls	8 simultaneous voice paths between the analog ports and the IP side (SIP phones, SIP trunks, IVR, conferences).
Call-progress tones	Dial, ring-back, busy and congestion tones come from the line cards themselves.
Management	A web browser on any PC on the network. No software to install, and no handset programming is needed.



Analog extensions are listed card by card as they are detected. Each port has its number, caller-ID name, groups and day/night class of service.

2. Numbering and dialling

- **Free numbering.** Every analog port, SIP phone, ring group, conference room and IVR takes a number of your choice. A number that is already used anywhere is refused, with a plain reason.
- **0 for an outside line.** 0 + number is checked before a line is taken. A mobile number goes out on a SIP trunk first and on an analog line only when no SIP trunk can take it. Other numbers go out on an analog CO line.
- **8N for a particular line.** 81 to 88 seize CO line 1 to 8 directly.
- **Caller identity codes (8001 and up).** Each SIP trunk's pilot number and its child numbers get their own access code, so a phone can call out showing a chosen company number.
- **Last number redial (*66).** Redials the last outside number the phone dialled, through the same checks as the first call.
- **Routed-only mode.** For strict sites, the 0 and 8N codes can be switched off so that every outside call goes through a configured route.
- **Number checker.** Type any number into the route page and see whether it stays internal, which route it takes, and what is actually sent to the provider.

- **Safe outbound routes.** A route cannot be saved if it would capture an internal number or make every internal call wait for more digits.

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3. Class of service and call control

- **Class of service, day and night.** Each phone has a day level and a night level: 0 Not connected, 1 Intercom only, 2 Local without 95, 3 Local with 95, 4 STD, 5 ISD. A call above the phone's level gets busy tone.
- **Abbreviated class of service.** A phone may dial Global Memory numbers even when its own level would refuse them.
- **Room-to-room calling switch.** Turn off all outgoing calls from a phone (guest rooms, lobby phones) while it still receives calls.
- **Automatic day/night changeover.** Day starts and ends at set times, with weekly holidays in night mode all day and a manual Day/Night override.
- **Call barring by number.** Bar numbers by their first digits (for example `00` for all international calls, or `1900` for premium-rate numbers), for all lines or for one SIP trunk or one CO line. Barring is checked against the number actually sent out.
- **Barring on direct line access.** Even on 8N, where the phone dials into the exchange's own dial tone, the system listens to the digits and drops the line as soon as a barred number is dialled.
- **Emergency numbers.** Allowed without an access code, whatever the class of service or barring. They go out on an analog line by default, so they keep working when the internet is down. (A phone with all outgoing calls switched off, the room-to-room setting, cannot dial them either.)
- **Allowed extensions per route.** Restrict a route (for example international) to named extensions.
- **Automatic call disconnection.** A maximum call length per route, with a warning before the call is cut.
- **Account codes (*50).** Enter a code before an outside call; it is stored in the call record for billing.

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4. Incoming calls

- **CO line ringing assignment.** Each analog line rings its own destination: an extension, ring group, IVR, conference or announcement.
- **Night service per line.** Each line has a night destination, used when the system is in night mode or during a chosen schedule.
- **Caller ID on CO lines.** FCT/DTMF, FSK type 1 and FSK type 2 caller ID decoded per line and shown on the ringing phones.
- **Caller ID on analog extensions.** The caller's number is sent to analog phones with a display.
- **DID routing on SIP trunks.** Route each dialled number to its own destination, by schedule, with a fallback destination outside hours.
- **CLI-based routing.** Send calls from chosen caller numbers to a chosen destination.
- **Ring groups (department call).** Ring all members together or hunt one after another, with a destination if nobody answers.

- **IVR / auto attendant.** Greeting with keypad options to departments, extensions, conferences or another menu; timeout and wrong-key handling.
- **Announcements.** Play a recording (for example an after-hours message) and then hang up or continue to a destination.
- **Schedules.** Office hours, holidays (date ranges) and custom schedules, used by routes and night service.

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5. Features at the handset

All of these work from an ordinary analog phone with a flash (recall) key, and from SIP phones. The codes can be changed or switched off on the Feature Codes page.

Feature	How it is used
Call transfer	Flash, dial the extension, hang up (or wait for an answer first). An unanswered transfer rings you back.
Call hold	Flash. Flash again to return.
Three-party conference	Flash, call the second party, flash again, then 7: all three in speech.
Call pickup	*8 answers a call ringing in your pickup group.
Directed pickup	758 + extension answers that ringing extension, from anywhere.
Call park and retrieve	Flash + *7 parks the call under your own number. 78 retrieves it; 78 + extension retrieves another's.
Auto call back	Press # on busy tone: when the busy extension is free it rings, then your phone rings.
Intrusion	766 + extension: a warning beep, then you speak to them while their party waits.
Silent barge-in	768 + extension: join the call, all three in speech. Can be switched off where not permitted.
Call forward	*72 + extension + # to set, *73 to cancel. SIP extensions can also have forward always, on busy and on no answer set from the web page.
Do not disturb	Set per SIP extension from the web page; callers get busy.
Global Memory	Dial a 2-digit location (20 to 99) to call a stored outside number.
Self number check	*11 speaks your own extension number.
Port number check	*12 speaks your extension, then your card and port number.
Self ring check	*13, hang up: the phone rings back as a bell test.
Echo test	*43 plays back what you say, to check the audio path.
Account code call	*50, account code #, number #.

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6. Conferences, music and recordings

- **Conference rooms.** Numbered rooms that callers dial into, with an optional PIN, an administrator PIN and a maximum number of members.
- **Music on hold.** Several music classes, with your own tracks uploaded from the browser.
- **Sound files.** Upload greetings and announcements (WAV 8 kHz mono, GSM, μ-law, A-law) for the IVR and announcements.

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7. Analog line quality

- **Echo cancellation** on every analog-to-SIP call, with an adjustable length.
- **Hardware bridging** of local analog calls in the line-card crosspoint, so internal calls between analog phones have no added delay or processing.
- **Line-card tones** for dial tone, ring-back, busy and congestion.
- **Power-on self-test** of the voice paths. A faulty path is taken out of service automatically.

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8. Monitoring and records

- **Dashboard.** Calls in progress, SIP phones online, SIP trunks registered, analog ports and analog lines at a glance.
- **Live calls.** Every call in progress, refreshed every 3 seconds, with a hang-up button.
- **Call records (SMDR / CDR).** Filter by date, caller, number dialled, line and result. Export to CSV for billing.
- **Analog hardware status.** Fitted line cards, phones off hook, voice paths in use, hardware-bridged calls and the health of the backplane link.
- **Event log.** Who changed what and when, sign-ins, and system events.
- **System logs.** The call engine's log, with a text filter.
- **Network tools.** Ping, traceroute and DNS lookup from the box itself.

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9. Administration and security

- **Three user roles.** Admin (everything), Manager (PBX configuration) and User (view only).
- **Forced password change** at first sign-in, and an automatic sign-out after a set idle time.
- **Login lockout.** An address that keeps failing to sign in is blocked for a while.
- **SIPguard.** Blocks internet addresses that try to guess SIP passwords. Office phones on the local network are never blocked.
- **Allow and deny lists** of addresses for web sign-in.
- **Bulk creation** of SIP extensions, with a random password for each.
- **SIP behind NAT.** External address and local network settings for trunks reached through a VPN router or firewall.

- **Network settings** (DHCP or static, static routes) with an automatic undo: if the new address cannot be reached within 90 seconds, the old settings return.
- **Date, time and NTP**, with Indian time zones, and a scheduled weekly reboot at a quiet hour.
- **Software update from the browser.** The new program is test-run before it is installed, and the previous version is kept.
- **Factory reset**, which keeps network settings so the box stays reachable.

Telematics IP-TCX. Features and limits are those of the software in service on 09/11/26 and may change with later releases.