PRODUCT DATASHEET

CallAnywhere PBX

Enterprise IP-PBX and contact centre, on infrastructure you own.

One platform for the whole telephone estate: carrier trunks, desk phones, browser and mobile softphones, IVR, call queues, a live supervisor wallboard, an agent workspace with CRM screen-pop and enforced wrap-up, call recording, and the reporting that turns all of it into numbers you can act on. Installed on a single Ubuntu server, administered entirely from the browser, with no per-seat licence and no third party in the media path.

1 server
Complete PBX and contact centre, one-command install

0 clients
Browser softphone — nothing to install on the desk

13+
Built-in contact-centre and CDR reports

100%
On-premise or your own cloud — your data stays yours

DOCUMENT	EDITION	ISSUED	CLASSIFICATION
Product Datasheet	**Complete platform**	**August 2026**	**Public**

01 Overview

Most organisations end up running a PBX for the phones, a separate contact-centre product for the queues, and a spreadsheet for everything neither of them reports on. CallAnywhere PBX is built as one system, so that a call, the agent who took it, the customer record it belongs to and the outcome logged against it are all the same record — not three systems that have to be reconciled afterwards.

Own the platform

Runs on your hardware or your cloud tenancy. Calls, recordings and customer data never leave the server. No per-agent subscription, no minimum term, and no vendor holding your numbers hostage at renewal.

Administered from a browser

Trunks, extensions, routes, IVR, queues, certificates, networking and backups are all configured in the panel. No configuration files to hand-edit and no shell access required for day-to-day operation.

Built on Asterisk

The open telephony engine carrying a large share of the world's SIP traffic, with the modern PJSIP stack — proven interoperability with effectively any SIP carrier, gateway or handset you already have.

WHO IT IS FOR

Contact centres

Inbound teams that need queues, live supervision, screen-pop, dispositions and wrap-up enforcement — with reporting that survives an audit.

Multi-site businesses

Organisations consolidating several PBXs, extensions across offices, and carrier trunks from more than one provider onto a single managed platform.

Service providers

Operators deploying a repeatable, brandable platform per customer, where a new site is a one-command install rather than a project.

IN THIS DOCUMENT

02 Platform capabilities

Everything listed here is included. There are no feature modules licensed separately.

Call routing & PBX core

- **Inbound routes** — pattern-matched DID routing to any destination
- **IVR menus** — multi-level, with uploaded or recorded prompts
- **Time conditions** — office hours, holidays, out-of-hours handling
- **Call queues** — strategies, priority, announcements, overflow
- **Outbound routes** with dial patterns and trunk failover order
- **Announcements** and **Music on Hold** managed in the panel
- One reusable destination picker across every routing screen

Trunks & connectivity

- **SIP trunk (registration)** — the PBX registers to your carrier
- **SIP trunk (IP / peer)** — the carrier authorises your address
- **Register-to-PBX** — GSM/SIM gateways and inbound-registering providers
- **WhatsApp Business Calling** via Meta, several numbers on one PBX
- Per-trunk codec lists, transports and caller-ID presentation
- Trunk destinations for onward routing and cascading failover

Extensions & endpoints

- **Browser softphone** — WebRTC over secure WebSocket, nothing to install
- **Mobile push-to-wake**, so a sleeping app still rings
- Any standards-compliant **SIP desk phone** or DECT handset
- **Call pickup** — `*8` group pickup, directed `**<ext>`
- **Boss / Secretary (PA)** call screening
- **Auto callback** (camp-on) when an extension is busy
- Live registration state, and a report of every register and drop

Contact centre

- **Live wallboard** — waiting callers, talking agents, hold times, second by second
- **Dynamic sign-in** — membership means "on the floor right now"
- **Listen, whisper and barge**, every use audit-logged
- **Pick a waiting call** out of a queue and hand it to an agent
- **Sticky agent** — repeat callers routed back to who they spoke to
- Pause reasons and break tracking, with handset codes `*45` – `*49`

Agent workspace

- **Screen-pop on ring** — the record is open before the agent says hello
- **Caller history strip** — past calls, recordings, what changed last time
- **Mandatory wrap-up** — form, then outcome, then the next call
- Dispositions with real behaviour: notes required, callbacks booked, success flagged
- **My callbacks**, owned by the agent who promised them
- Day statistics that add up: talk, wrap, break and idle across the shift

Built-in CRM

- **Drag-and-drop form builder** — design the screen your agents actually need
- **Customers and services** — one customer, many lines, policies or accounts
- Forms mapped per queue, so support and sales capture different things
- **CSV list import** with column mapping and a dry-run preview
- Full lead history — who changed what, on which call

Voice broadcast

- Announcement campaigns to an imported list — no agent required
- Scheduled dialling windows with per-campaign caller ID
- Per-call result recorded and reportable
- Payment reminders, outage notices, appointment prompts

Reporting & analytics

- **Call records (CDR)** with recording playback and search
- Queue summary, agent summary, agent activity, agent sessions
- Abandoned calls, hold time, hourly interval, dispositions, wrap-up
- Callbacks, sticky-agent effectiveness, extension registrations
- **CSV export** on every report, hardened against formula injection
- Optional **Grafana** dashboards over a dedicated event store

Administration

- **Users, roles and granular permissions** — per module and per item
- **Network settings** from the GUI, with automatic rollback if you lock yourself out
- **SSL** certificates, **SMTP** mail, firewall and IP allow-listing
- **Maintenance mode**, system status and live service health
- **Data cleanup** — archive first, delete second, restore if you were wrong
- White-label branding: your company name on the sign-in screen

Recording & retention

- Call recording with in-browser playback from the call record
- Access governed by permission, not by knowing the file name
- Retention windows with archive-before-delete
- Archived data exported as newline-delimited JSON, restorable

Agent assist

- Mirror live call audio to your own analytics or AI service
- Policy controlled from the panel, and **off by default**
- Asterisk streams directly — the panel is never a data channel
- Health reporting, so a silently unconfigured feed is visible

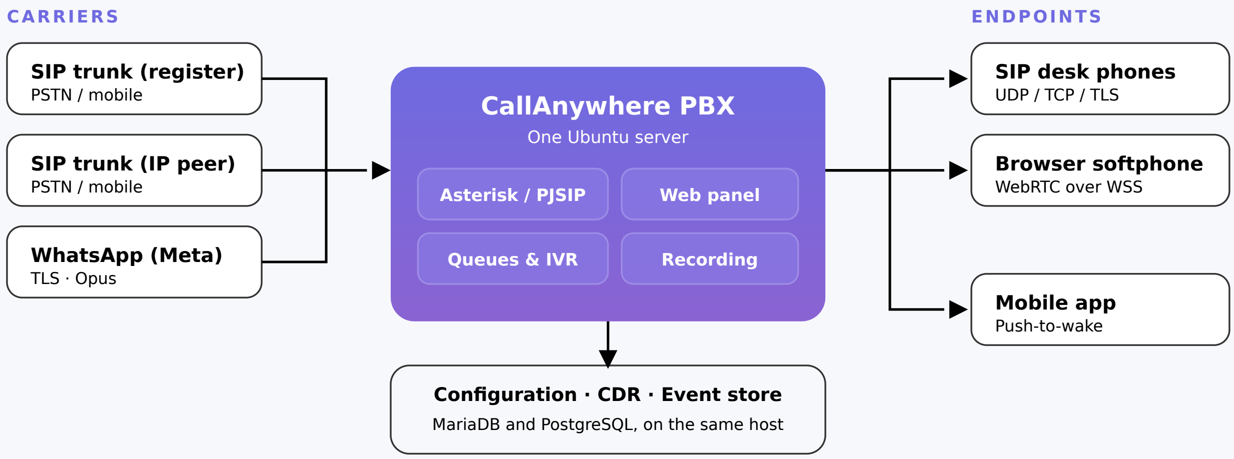
Directory & self-service

- Company directory, searchable from every screen
- **My Phone** — the user's own extension, calls and settings
- Personal call history and scheduled callbacks
- Role-scoped accounts, so a manager sees a team and not the estate

Permissions are default-deny. A role sees a screen only if it has been granted, and grants can be scoped to specific queues — so a sales team leader reads sales numbers and cannot read support's from another tab. Every agent, however, gets their workspace with no configuration at all: sign in, take calls, log the outcome. That is a deliberate guarantee, not a side effect of the default role.

03 Architecture

A single server carries signalling, media, the web panel and the databases.



Signalling and media terminate on your own server. Nothing about a call — audio, recording, caller identity or CRM record — is relayed through a third-party service.

04 Security & compliance

Call records are personal data. The platform is built and audited on that basis, with controls aligned to CERT-In expectations for logging, retention and incident readiness.

- **TLS 1.3 with AES-256-GCM** on SIP, WebRTC and the panel alike
- **HTTPS everywhere**, with HSTS and managed certificates
- **Content Security Policy** and a full set of hardening headers
- **Audit trail** of supervisor monitoring — who listened to whose call
- **Retention and purge** — archive, delete, and restore if needed
- **Administrator recovery** from the console without weakening the login
- **SRTP voice encryption** — AES-256 suites on SIP/TLS endpoints
- **Default-deny permissions**, enforced at every API endpoint
- **Rate limiting** and **fail2ban** against credential stuffing and SIP scanning
- **Recording access control**, so agents hear only what they are entitled to
- **No archives served** from the web root, enforced by the server itself

Independently exercised. The platform has been through a VAPT pass in which legacy endpoints were removed rather than patched, publicly reachable exports were withdrawn from the web root, and server-level backstops were added so the same class of mistake cannot return unnoticed in a future deployment.

05 Technical specifications

Reference configuration as shipped. Individual components are version-pinned per release.

PLATFORM

Telephony engine	Asterisk 20 LTS, PJSIP channel driver
Operating system	Ubuntu Server 24.04 LTS (64-bit)
Web tier	Apache 2.4 with PHP 8.3
Databases	MariaDB 10.11 — configuration, call records, CRM PostgreSQL 16 — real-time telephony event store
Real-time services	Node.js event recorder with WebSocket push to the panel
Optional analytics	Grafana dashboards, reverse-proxied under the same hostname

INTERFACES & PROTOCOLS

SIP signalling	UDP, TCP and TLS — TLS 1.3 with **AES-256-GCM**
Browser telephony	WebRTC over secure WebSocket (WSS), carried on TLS 1.3 with **AES-256-GCM**; media keyed by DTLS-SRTP, with ICE for traversal
Voice encryption	SRTP on every encrypted leg — **AES-256** suites (AES-256-CM-HMAC-SHA1-80, AEAD-AES-256-GCM) on SIP/TLS endpoints; browsers use the AES-128 DTLS-SRTP profile that the WebRTC standard mandates
Audio codecs	G.711 μ-law and A-law, G.729, Opus — selectable per trunk and per extension
DTMF	RFC 2833 / RFC 4733, SIP INFO, in-band
Carrier types	Registration-based SIP, IP-authorised peer, inbound-registering gateway, WhatsApp Business Calling
Recording formats	WAV and GSM, played back in the browser
Exports	CSV from every report; archived data as newline-delimited JSON

REQUIREMENTS

Server — minimum	4 vCPU, 8 GB RAM, 100 GB disk
Scaling	No imposed limit on agents, extensions or concurrent channels — capacity is bounded only by the hardware allocated, so anything above the minimum simply carries more
Network	Static public address or 1:1 NAT; a dedicated interface for the SIP trunk is supported
Bandwidth	Approximately 90 kbit/s per concurrent G.711 call, each direction
Agent workstation	Any current Chrome, Edge, Firefox or Safari, with a USB or Bluetooth headset
Certificates	A public TLS certificate for the PBX hostname — required by the browser softphone

06 Deployment & support

Designed so that a second site, a staging system or a customer's own server is a one-command job.

One-command install

A single command on a clean Ubuntu server brings up Asterisk, the databases, the web panel, the background services and the firewall rules — already configured and talking to each other.

Survives a move

A change of IP address or hostname is re-derived automatically, so a migration between clouds, or a DHCP lease that changes underneath you, does not silently break registration overnight.

Reversible by design

Backups are taken before changes that matter, network edits roll themselves back if you lose your session, and deleted data is archived before it is removed.

DEPLOYMENT OPTIONS

On-premise	Your own hardware, in your own rack, on your own network
Virtualised	A virtual machine in your existing hypervisor
Cloud	An instance in the region and jurisdiction you choose
Edition	The same complete build in every case — there is no cut-down edition and no feature paywall

The minimum is a floor, not a ceiling. The platform imposes no licensed limit, so capacity is a hardware question: allocate more and it carries more. What actually governs concurrency is codec choice, recording and transcoding rather than agent count — G.711 end-to-end with no transcoding carries substantially more simultaneous calls on the same server than a deployment transcoding every leg. Size disk against your recording retention period, which is usually the fastest-growing figure.

GETTING STARTED

01

Scope

We size the server against your real call volumes, review the carrier arrangements you already have, and agree what moves across first.

02

Install

One command on a clean Ubuntu server. Trunks, extensions, routes, queues and your CRM forms are configured with you, not handed over as homework.

03

Go live

Point or port your numbers, train supervisors and agents on the live system, and run in parallel with the old platform until you are happy.

CALLANYWHERE PBX

Ready when you are.

We are happy to size a system against your real call volumes, review your carrier arrangements, or set up a working trial you can put on a live queue. No obligation, and no sales process to sit through first.

WEB
callanywhere.co.in

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DEMO
Ask for guided access to a live system

- A complete PBX and contact centre in one platform
- Your server, your data, no per-seat licence
- TLS 1.3 and SRTP on every leg that carries voice
- Browser softphone — nothing to install on the desk
- Installed and running from a single command
- Thirteen built-in reports, all exportable

DOCUMENT
Product Datasheet

ISSUED
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