

SOLUTION BRIEF

From Analog & ISDN Migration to SIP Trunking *A Smooth, Secure and Sustainable Path to IP Voice*

The End of the PSTN Era — But Not Everywhere at Once

Public switched telephone networks are being switched off across the world, but not at the same pace and nowhere overnight. Some countries have completed the national transition, and some have moved its subscribers over. Some even stopped selling analog lines and are closing the installed base in stages, region by region, while some are running a national migration program whose completion date has already been revised.

In many other markets — and in the more remote parts of markets already committed to switch-off — copper is still carrying live traffic today.

The direction of travel is settled: copper-based voice is being retired in favor of VoIP and cloud-based communications. The timetable is not.

For most service providers and enterprises, the realistic outlook is a long overlap, during which SIP trunks, ISDN accesses and analog lines coexist across the estate — sometimes within the same building.

Session Initiation Protocol (SIP) trunking is the technology that replaces it. Since its introduction in 2005 it has become the industry standard for the next generation of unified business communications, and the availability of reliable high-speed broadband has accelerated adoption sharply. SIP offers compelling advantages over traditional PBX-based telephony — multimedia streaming, conferencing, real-time collaboration — generating significant cost savings and flexibility for small businesses and large enterprises alike, and opening the door to UC, UCaaS and UCC services.

What makes this migration difficult is not the telephone. It is everything else connected to the old network: monitored fire alarms, lift emergency lines, telemetry and payment terminals. Many of these cannot move to IP without replacing every remote terminal and central collector — an expense rarely budgeted for.

So the pure analog part of the migration remains unfinished almost everywhere. The pragmatic answer is neither to wait for it nor to force it, but to separate the two questions: move the transport to IP now and leave the analog and ISDN terminals in place until they are renewed, site by site, on the customer's own budget and schedule.

That is what the Ekinops portfolio is built to do — from a two-port analog terminal adapter on an isolated lift line or alarm panel, up to multi-PRI aggregation for a 300-user branch — and, when the terminals themselves finally move to IP, the same platform becomes the enterprise session border controller that the all-IP site needs.

► Why SIP Trunking Wins

- Unlimited connections and voice sessions, subject to bandwidth
- Voice, video, presence, conference and chat on one service
- Needs only a low-cost eSBC connected to an internal IP-PBX
- More efficient use of access bandwidth
- Cost reductions in excess of 50% are achievable
- Improved employee productivity

The Migration Continuum — From Analog Backhaul to All-IP

Leaving the PSTN behind is not one project but two, usually separated by several years. The first moves the transport to IP while the terminals stay as they are. The second moves the **terminals** themselves — the PBX gives way to a hosted phone system, desks move to SIP or Microsoft Teams, and the analog devices are finally replaced. Most enterprises live somewhere between the two for a long time. The question that decides the cost of the whole programme is whether the equipment installed for the first project can still serve the second.

► Phase 1 — Analog and ISDN carried over IP

The PBX, the phone system, the alarm panels, lift lines, fax and payment terminals stay exactly as they are. The Ekinops router presents FXS, BRI or PRI interfaces locally and carries voice, DTMF, fax (T.38 or pass-through) and modem signaling transparently over SIP to the operator network or the alarm collector. Nothing changes for the end user — and this is where the job of a pure voice gateway ends.

► Phase 2 — A mixed estate, gateway and SBC at the same time

The first IP endpoints arrive: a hosted PBX or UCaaS seat, Teams for head-office users, SIP phones in a new building — while the legacy PBX and the analog terminals are still in service. The site now needs both roles at once. Registration and authentication of the IP phones, media anchoring, topology hiding, transcoding between incompatible codecs and NAT traversal become daily requirements, alongside the FXS and ISDN ports that are still carrying alarms.

► Phase 3 — The all-IP site

The last analog terminal is replaced and the PBX is retired. The router's role is now purely that of an enterprise session border controller: demarcation between the enterprise and the provider's SIP network, security, interoperability with whichever SIP flavor the platform speaks, call admission control and on-premise survivability. The remaining analog and ISDN ports become a backup path rather than the main service.

The same platform covers all three phases. ONeSBC is a software license on the Ekinops routers, so moving from phase 1 to phase 3 is a license and a configuration change — not a new box, a new installation visit or a new management system.

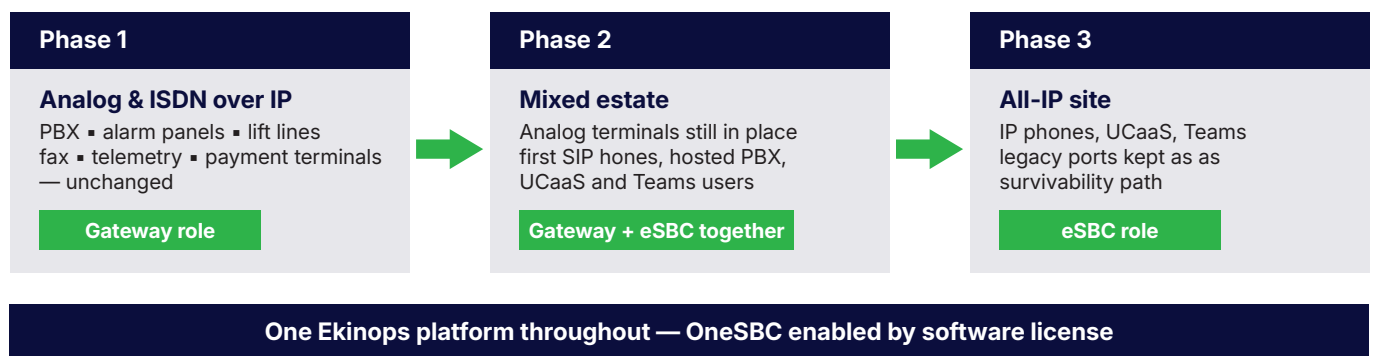


Figure 1 - The migration continuum: the same CPE serves the analog estate, the mixed estate and the all-IP site

Where a Voice Gateway Stops — and a Voice Router Continues

This is the practical difference between a voice gateway and a voice router. A pure gateway is built for one job: presenting analog or ISDN interfaces to legacy terminals and converting them to SIP. It does that job well, and then it stops. The moment the customer starts moving terminals to IP — which is the moment the program actually becomes a voice-over-IP project rather than an access migration — the gateway has no role in the new architecture. It has to be replaced, together with the installation visit, the provisioning templates and the operational tooling built around it.

An Ekinops voice router carries the same interfaces but adds the enterprise session border controller that phases 2 and 3 require. The investment made for the analog cutover is the investment that serves the all-IP site.

Requirement	Pure voice gateway	Ekinops voice router with ONeSBC
Keep analog and ISDN terminals in service	Yes	Yes — FXS, BRI and PRI with fax, DTMF and modem transparency
Serve SIP phones, hosted PBX and UCaaS seats	Outside its scope	SIP Centrex mode with phone authentication, registration and media anchoring
Microsoft Teams Direct Routing	Outside its scope	Certified
Security demarcation toward the SIP trunk	Outside its scope	Topology hiding, IP-level filtering, anti-flooding, DoS self-protection, NAT traversal
Interworking between SIP flavours	Limited	SIP header manipulation, SIP Connect 2.0, media, DTMF and fax transcoding
Service continuity if the WAN fails	Outside its scope	On-premise disaster recovery with local call control and fallback over ISDN or analog trunks; HA in master/slave or active/standby
Data, LAN, SD-WAN and IP-VPN on the same box	Outside its scope	Included — one CPE, one management system, one contract
Path to the all-IP site	Replace the hardware	Add a software license to the installed hardware

How Ekinops Can Help

► Flexible and Extensible

Ekinops advanced branch office router solutions give communication service providers a highly flexible and extensible SIP trunking service platform, as well as carrier-grade Internet access devices supporting a broad range of business use cases.

► Granular and Scalable

By supporting SIP over TCP, TLS and SRTP, Ekinops routers deliver cost-effective enterprise SBC services — ONeSBC — for managed voice delivered in parallel with high-speed xDSL and fibre Internet access, giving maximum resilience and reliable performance for SMEs and distributed, multi-location businesses.

► Cost Effective

Combining enterprise-class unified communications with LAN, WAN and cloud service access lets providers offer secure, value-for-money bundled services that maximize return for both the customer and the provider. Ekinops SIP-based CPE devices give providers a strong value proposition, enabling their customers to plan a controlled and budgeted transition from legacy PBX technology to the latest IP-based communications systems, in line with the ISDN phase-out.

ONeSBC — Session Border Control Inside the Router

ONeSBC is the enterprise SBC software option for the ONE Series multi-service access routers. It provides signal and media interworking, address translation, network demarcation, quality of service and security, and it is what allows the same CPE to follow the customer from the analog estate to a cloud-based UCaaS phone system.

► Two modes of operation

In *SIP Centrex* mode, optimised for hosted services, the SBC manages SIP phone authentication and registration to the service provider's network, anchors media so that the RTP stream follows the optimum path, and provides call survivability should the connection to the SIP application server or hosted PBX become unavailable. In *SIP Trunking* mode, a back-to-back user agent ensures interoperability between SIP-based PBXs and the provider network, with transcoding where media is incompatible. Selective transcoding releases DSP resources when no transcoding is required, improving scalability.

► Interoperability without an integration project

ONeSBC translates between the various SIP implementations found on SIP-based PBXs, SIP phones and the provider's network, giving plug-and-play behaviour for Centrex and trunking services and avoiding the engineering cost of larger-scale SBC platforms for SME sites. Media interworking and transcoding for voice, DTMF and fax calls is precisely the mechanism that lets analog services converge with IP ones during phase 2.

► Security at the border

Acting as the border between the customer premise and the provider network, ONeSBC hides the topology between IP networks, securely manages SIP sessions, NAT translation and firewall traversal and SIP phone registration, and protects itself against denial-of-service attacks.

► On-premise disaster recovery and high availability

ONeSBC detects network failure automatically and assumes local call control: internal on-site calls continue, and inbound and outbound calls can be delivered over the alternative ISDN and analog voice trunks on the same router. Once the trunk network is restored, control is handed back to the provider's network. This is where the legacy interfaces earn their keep a second time — the ports installed for phase 1 become the survivability path of the all-IP site. High availability strategies cover both a redundancy topology with two SBC devices connected to the WAN/IMS network and failover where the SBC has multiple WAN/IMS connectivity, in master/slave or active/standby mode, with calls maintained during failover.

► Lower total cost of ownership

Integrating SBC and SIP proxy capabilities within a single multi-service access router reduces capital cost, logistics and operations. For SIP trunking, no unwanted SIP header, SDP or media is sent to the provider network, avoiding operational and support issues.

Ready for Microsoft Teams

ONeSBC natively supports Microsoft Direct Routing, with secure connectivity to the Microsoft Phone System platform, and is certified for Teams deployment on the ONE526L, ONE526, ONE1526 and ONE2515. For most enterprises Teams is the trigger for phase 2: head-office users move to the cloud platform while the branch alarms and lift lines are still analog. The same router serves both on the same access, with call admission control and quality of service protecting the critical flows.

Recommended Platforms



ONE526L

Intelligent 2-FXS Analog Terminal Adaptor

- 2 FXS ports with DSP for 8 voice channels, REN 3 — the entry point for an isolated analog terminal
- Datasheet at: <https://www.ekinops.com/products-services/products/oneaccess/voice-data-routers/voice-routers>



ONE526

Hybrid Access Branch Office Router

- BRI SIP voice and data router with 8 BRI, 12 FXS, 1 FXO
- Datasheet at: <https://www.ekinops.com/products-services/products/oneaccess/voice-data-routers/voice-routers>



ONE1526

Hybrid Access Branch-Office Router for Converged Services

- 1 or 2 PRI E1/T1** and up to 18 FXS / 1 FXO — sized for sites of up to 300 users mixing ISDN, FXS and IP
- Datasheet at: <https://www.ekinops.com/products-services/products/oneaccess/voice-data-routers/voice-routers>



ONE2515

Next Generation High Speed Multi-service Access Router

- 2 or 4 PRI SIP voice and data router with 24 FXS, 2 FXO
- Datasheet at: <https://www.ekinops.com/products-services/products/oneaccess/voice-data-routers/voice-routers>

Choosing the Right Platform

Platform	Analog interfaces	ISDN	WAN access	Typical site
ONE526L	2 FXS	—	A/VDSL2, Gigabit Ethernet, 4G/LTE	A single isolated terminal: lift line, alarm panel, fax, payment terminal
ONE526	12 FXS, 1 FXO	8 BRI	A/VDSL2, 1GE UTP/SFP, 4G/LTE	Small branch mixing analog terminals and ISDN
ONE1526	Up to 18 FXS, 1 FXO	1 or 2 PRI E1/T1	A/VDSL2, 1GE UTP/SFP, 4G/LTE	Branch office of up to 300 users, converged services
ONE2515	24 FXS, 2 FXO	2 or 4 PRI	Dual GE UTP/SFP, 4G/LTE	Large site, or the collector end of the alarm chain

All four platforms run OneOS6, support SIP over UDP/TCP/TLS with RTP/RTCP, and offer the ONeSBC enterprise session border controller under license.

ONeSBC Capacity and Licensing

Per platform (running OneOS6)	ONE526 ONE526L	ONE1526	ONE2515 2P	ONE2515 4P
SIP phones (Centrex seats)	200	200	1200	1200
Simultaneous Centrex calls (no transcoding)	100	100	600	600
Maximum number of SIP trunks	200	200	200	200
Simultaneous trunking calls (UDP/TCP)	100	100	600	600
Simultaneous calls with TLS / SRTP	60	60	400	400
Simultaneous transcoded trunking calls	8	15	35	70
Available ONeSBC licenses (simultaneous calls)	10 / 25 50 / 100	10 / 25 50 / 100	50 / 100 200 / 600	50 / 100 200 / 600

The «ONeSBC-x» license activates all ONeSBC features for up to x simultaneous calls, within the maximum number of calls the selected service can perform on that platform.

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About Ekinops

Ekinops is a European leader in optical networking and secured connectivity solutions for service providers, enterprises and public organizations around the world.

Ekinops designs, develops and manufactures high-performance solutions with cutting-edge software used to deliver high-bandwidth connectivity and managed services in over 70 countries worldwide.

Ekinops leads in the areas of:

- Advanced coherent optical networking for all metropolitan, regional, long-distance and datacenter interconnection applications
- Secured connectivity solutions for enterprises, from voice and data routing devices to Secure Access Service Edge (SASE) capabilities, including ZTNA (Zero-Trust Network Access), SD-WAN, and SSE.

Ekinops operates globally and is listed on Euronext Paris (FR0011466069 - EKI).