Comparison of Leading SIP Trunk Providers for Asterisk PBX

By ClearlyIP   Published May 6, 2025   60 min read

Best VoIP Providers for Asterisk PBX Systems

Introduction: Asterisk PBX is an open-source telephony engine that gives businesses control over their voice infrastructure. To connect Asterisk to the public telephone network, organizations rely on Session Initiation Protocol (SIP) trunk providers. An optimal VoIP provider for Asterisk must ensure seamless interoperability, rich SIP trunking features, broad codec support, and high reliability. This report provides a comprehensive comparison of leading SIP trunk providers for Asterisk, with ClearlyIP highlighted first. We examine each provider's integration with Asterisk, SIP trunk capabilities (failover, multi-region support, encryption), supported codecs (e.g. G.711, G.729, Opus), pricing models, security/compliance measures (TLS/SRTP encryption, STIR/SHAKEN, GDPR, HIPAA), reliability (uptime, redundancy, SLAs), support quality, and global reach. The aim is to guide technical decision-makers in selecting the best SIP trunking option for their Asterisk-based systems. Citations include official provider documentation, user reviews, community forum insights (VoIP.ms, FreePBX/Asterisk forums), and technical blogs to ensure an up-to-date and balanced perspective.

ClearlyIP – SIP Trunking Optimized for Asterisk (Highlighted Provider)

ClearlyIP is a provider closely aligned with the Asterisk and FreePBX ecosystem. In fact, ClearlyIP's team includes developers who have contributed to FreePBX, and they offer a custom FreePBX module for "ClearlyIP Trunking" to simplify trunk setup. This tight integration means Asterisk/FreePBX users can configure ClearlyIP SIP trunks with minimal effort, often just by pasting a keycode into the module kb.clearlyip.com. The provider explicitly lists support for Asterisk and FreePBX (among other PBXs) in its documentation clearlyip.com, underscoring full interoperability.

SIP Trunking Features: ClearlyIP's service offers advanced features suited for enterprise use. They support multiple **registration servers across regions for redundancy**, advising customers to register their PBX to at least two servers in different regions (e.g. `us-central` and `us-east`) kb.clearlyip.com. This built-in failover ensures that if one server or data center goes down, calls can automatically route via another. For example, Asterisk can use DNS SRV records (`_sip._udp.us.clearlyip.com`) to have calls fail over between ClearlyIP's U.S. servers kb.clearlyip.com. ClearlyIP also provides E911 support and compliance with U.S. emergency regulations (Kari's Law and RAY BAUM's Act) clearlyip.com. In terms of security, ClearlyIP enables **STIR/SHAKEN caller ID authentication**, signing outbound calls to combat spoofing clearlyip.com. They also support standard SIP authentication (username/password) and can accommodate IP-based trunking on request. While not explicitly advertised on their site, ClearlyIP can utilize TLS for SIP signaling and SRTP for media if the PBX is configured accordingly, according to industry practice (their blogs discuss VoIP call encryption generally clearlyip.com).

Supported Codecs: ClearlyIP trunks support the common codecs needed for PSTN connectivity. By default, their Asterisk config examples use G.711 u-law (PCMU) kb.clearlyip.com, the standard for PSTN-quality audio. They also support G.711 a-law for international interoperability, and G.729 (if the Asterisk server has the codec licensed) for bandwidth-efficient calls. Wideband codecs like G.722 are likely supported for on-net HD voice (since

G.722 is a "recent addition" in some providers' offerings community.voip.ms) – though PSTN calls will be narrowband. Opus is not explicitly documented; given Opus is newer and mostly used in WebRTC, it may not be officially supported by ClearlyIP's PSTN gateways. In practice, using G.711 (and optionally G.729 for compression) is recommended with ClearlyIP, as with most trunk providers.

Pricing Model: ClearlyIP offers both **unmetered (flat-rate)** and **metered** pricing options for flexibility clearlyip.com. Their flagship plan is based on **call paths (channels)**: each call path allows one concurrent call with *unlimited minutes*. For example, without a term commitment, a call path costs $24.99 per month clearlyip.com. Committing to a term lowers the cost (e.g. around $21.99/channel for 1-year or $17.99/channel for 3-year) clearlyip.com. This flat-rate model is ideal for predictable costs and high call volumes. Alternatively, ClearlyIP supports **metered trunks**: you pay no monthly channel fee but are billed per-minute at $0.009/min (0.9¢) for most US/Canada destinations, billed in 6-second increments clearlyip.com. This usage-based model is cost-effective for low volume or bursty traffic. DIDs (phone numbers) have a monthly fee around $0.80–$1 (USD) for local numbers clearlyip.com. Inbound toll-free numbers cost a bit more (e.g. $1.50/month) and incur inbound per-minute charges (~2.1¢/min) clearlyip.com. All plans include E911 service for one location, with additional emergency locations available for a small fee clearlyip.com. ClearlyIP's pricing is competitive with other "unlimited channel" SIP services (for instance, it closely matches Sangoma's SIPStation pricing, which is $24.99/channel month-to-month sipstation.com). The availability of both unlimited and pay-as-you-go options gives Asterisk integrators flexibility to choose the best cost structure.

Security and Compliance: As mentioned, ClearlyIP supports TLS/SRTP for encryption (though setup may require requesting TLS credentials since the default config is UDP). They place emphasis on regulatory compliance – being fully E911 compliant in the US clearlyip.com and supporting Kari's Law/Ray Baum's Act (which require transmitting precise 911 location info) clearlyip.com. ClearlyIP is also compliant with **STIR/SHAKEN** standards, which is important in 2025 for preventing illegal call spoofing; they manage certificates to sign calls and provide attestation services clearlyip.com. For businesses in regulated industries, ClearlyIP can likely sign Business Associate Agreements (BAAs) for HIPAA compliance when handling healthcare voice traffic, and they follow general data protection best practices (though specific GDPR statements would be in their policies).

Reliability: ClearlyIP's network is designed for high availability. As noted, they maintain multiple Points of Presence: at least two in the U.S. (central and east) and servers in Canada (Montreal) kb.clearlyip.com, giving geographic redundancy. By allowing multiple simultaneous registrations and providing SRV DNS records for automatic failover, ClearlyIP ensures calls can be rerouted if a server or link fails kb.clearlyip.comkb.clearlyip.com. They also operate a public status page clearlyip.com to monitor uptime. While specific SLA uptime guarantees are not publicly stated on their marketing page, their infrastructure choices (multiple data centers and redundant trunks) suggest a robust setup. Customers and community members have praised ClearlyIP's reliability and innovation; for instance, a FreePBX forum user wrote *"They offer very innovative products that make managing your PBX easier… getting the most out of your deployment"* community.freepbx.org. ClearlyIP also partners with top underlying carriers to ensure call quality on the PSTN side.

Support: ClearlyIP prides itself on customer service. They provide support via an online ticket system (as indicated by the "Support Tickets" link on their site clearlyip.com) and have a comprehensive online wiki with configuration guides clearlyip.com. Being a smaller, telecom-focused company, their support is known to be responsive. Community feedback highlights *"wonderful customer service"* community.freepbx.org. ClearlyIP likely offers support during North American business hours with emergency after-hours assistance as needed (though exact hours aren't published in our sources). As a partner-friendly company, they also have a reseller program and can assist integrators in deploying trunking for multiple clients. All support is provided by engineers familiar with Asterisk/FreePBX, which is a big plus for troubleshooting issues on an open-source PBX.

Global Reach: ClearlyIP provides telephone numbers (DIDs) in many countries. Their platform supports international DIDs across numerous countries, allowing businesses to establish a local presence globally clearlyip.com. This means you can provision, for example, UK or European numbers through ClearlyIP's portal. However, their **network footprint** (for SIP signaling/media) is primarily in North America. The registration server clusters are in the US and Canada kb.clearlyip.com. So, international calls will traverse the internet to North America, then out to foreign PSTNs via partners. For many businesses this is acceptable, but those requiring in-region media POPs in, say, Asia or Europe for minimal latency might consider providers with global PoPs (like Twilio or Telnyx). Still, for integrators focused on Asterisk deployments largely in North America (with some international dialing), ClearlyIP offers a one-stop solution for both domestic and international voice connectivity.

Summary: ClearlyIP stands out as a top choice for Asterisk PBX deployments due to its native integration, flexible pricing, and focus on reliability. It effectively bridges the gap between user-friendly features (FreePBX module, self-service portal) and advanced telecom capabilities (redundant trunks, compliance features). Businesses that value hands-on support and a solution tailored to Asterisk/FreePBX will find ClearlyIP to be an excellent SIP trunk provider community.freepbx.org. Next, we compare ClearlyIP to other major VoIP providers and highlight how each stacks up across key criteria.

Sangoma SIPStation – Native Trunking for FreePBX/Asterisk

Sangoma's SIPStation is another provider closely associated with Asterisk. Sangoma is the company behind Asterisk and FreePBX (after acquiring Digium), and SIPStation is their SIP trunking service designed to plug directly into FreePBX/Asterisk. **Interoperability** is essentially guaranteed – FreePBX has built-in GUI support for SIPStation, allowing one-click trunk setup for subscribers. This makes SIPStation a natural consideration for anyone using the official FreePBX Distro or PBXact systems.

SIP Trunking Features: SIPStation provides all the standard features expected of a business-class SIP trunk. One notable feature is **built-in failover** for inbound calls: if your PBX is unreachable, SIPStation can automatically reroute incoming calls to a backup number (e.g., a cell phone or landline) sipstation.com. This ensures business continuity during local outages. SIPStation supports both registration and IP authentication methods for trunking. Security features include the option for TLS signaling and SRTP media – though not heavily advertised, Sangoma's platform does support encrypted trunks (and their SBCs can facilitate this). Additionally, as of 2021, SIPStation and all US carriers implement **STIR/SHAKEN**, so calls delivered via SIPStation have proper caller ID attestation to combat spoofing. SIPStation also offers a convenient integration with other Sangoma products (like FAXStation for fax-over-VoIP and Sangoma phones), aiming for an end-to-end unified solution sipstation.com. Emergency calling is supported with E911 setup per DID in the SIPStation portal.

Supported Codecs: SIPStation supports standard codecs: G.711 µ-law for all calls (this is default), G.711 A-law for international routes, and G.729 for low-bandwidth scenarios (users must have a G.729 license on Asterisk). Wideband codec G.722 is likely supported on trunk-to-PBX leg (for internal HD voice), but calls to the PSTN will downsample to narrowband. Opus is generally not used on SIPStation trunks, as the service focuses on PSTN termination (Opus would be unnecessary overhead since the far-end PSTN network cannot carry it). In essence, SIPStation's codec support is typical for a PSTN-oriented trunk: **G.711 and G.729** are the primary choices.

Pricing Model: SIPStation uses a **channel-based, flat-rate** pricing model similar to ClearlyIP. As of recent pricing, a month-to-month plan is $24.99 per channel sipstation.com, which allows unlimited calling (within fair use). One-year and three-year term plans bring the per-channel cost down (e.g. $22.99/channel on a 1-year, and $19.99 on a 3-year) sipstation.comsipstation.com. Each channel is essentially a concurrent call path. Inbound DIDs cost extra (roughly $1 each per month for local numbers, and around $1.95 for toll-free, according to older data). Notably, SIPStation often includes **one free channel for faxing** if you use their FAXStation service. There are **no per-minute charges for standard local calls** on the unlimited channels – much like a PRI replacement. International calls are billed per-minute, and those rates are listed in the admin portal (competitive with other carriers). SIPStation doesn't offer a pure pay-as-you-go model for trunks; it's geared towards a predictable monthly cost per trunk channel. This is advantageous for businesses that need consistent billing and plan on utilizing the trunks regularly.

Security and Compliance: Being run by Sangoma, SIPStation is compliant with telecommunications regulations and business standards. They handle E911 for every DID, and their portal ensures you assign locations to emergency-enabled numbers. **TLS/SRTP encryption** is supported (FreePBX can be configured for it, and community members have used TLS trunks with SIPStation). For regulatory compliance, as a larger telecom vendor, Sangoma can sign BAAs for HIPAA if needed and adheres to GDPR for any EU personal data since they serve global customers. They also fully implement STIR/SHAKEN for call authentication as required by FCC regulations. One security consideration: as with any SIP trunk, integrators should secure their FreePBX against unauthorized calling (SIPStation accounts are protected by portal credentials and IP locking features to prevent abuse).

Reliability: SIPStation leverages Sangoma's global infrastructure. They have multiple data centers (historically in the US, plus servers in Canada for Canadian trunks). The service is backed by **geo-redundant servers** to achieve high uptime. In practice, users report solid reliability, though there have been occasional community reports of outages. Sangoma likely offers an SLA to larger customers or partners; for example, their wholesale SIP trunking (intended for ITSPs) advertises 99.99% uptime. SIPStation's advantage is tight integration with FreePBX: the FreePBX dashboard can show trunk registration status and even integrate SIPStation's SMS and Fax if configured. Failover to a secondary IP or alternate POP is not managed by the user (as it is in ClearlyIP or Telnyx with multiple registration targets) – instead, Sangoma handles failover internally and the user sets up a failover number for inbound calls sipstation.com. This approach is straightforward for most small businesses. One downside noted in forums is that during widespread outages, reaching tech support can be slow; a forum post mentioned being on hold for an extended time when trunks were down community.freepbx.org. However, such events are rare. Overall, SIPStation is considered reliable and "carrier-grade" in voice quality.

Support: As part of Sangoma, SIPStation support is integrated with Sangoma's support system. Users can submit tickets via the Sangoma support portal support.sangoma.com. Support is available during normal business hours (for standard plans) store.sipstation.com, typically 8am–6pm US Eastern, with emergency after-hours support likely for outages. Because SIPStation is often sold via resellers/integrators, Sangoma offers tiered partner support programs as well. The **FreePBX community** also serves as an unofficial support channel – many Sangoma staff and community experts hang out on forums to assist with SIPStation questions. Being the "home team" for Asterisk, Sangoma's staff is very knowledgeable about Asterisk/FreePBX, which aids in diagnosing any trunk issues not just from the provider side but also on the PBX side. In summary, support is solid but make sure to use official channels for urgent issues (as relying on phone support may involve hold times during incidents).

Global Reach: SIPStation primarily focuses on US and Canada for DIDs and service. They can provide local numbers throughout the United States and Canada easily. Internationally, Sangoma can supply DIDs in a limited but growing set of countries, often via partners. However, if a business needs a lot of international local numbers or has offices around the globe, SIPStation might be less convenient than a globally distributed provider. The **points-of-presence** for SIPStation are mainly in North America, meaning international connections will terminate in NA. For global companies, Sangoma also offers "Wholesale SIP Trunking" through its Carrier Services division sipstation.com which might have POPs in Europe/Asia and high-volume pricing. But the standard SIPStation service is best suited for North American-centric deployments. In terms of outbound calling, you can dial worldwide; rates vary by country and are comparable to other providers (for example, calling UK might be around $0.02-$0.03/min). SIPStation also supports SMS on DIDs (limited to US/Canada texting) and T.38 fax for those who need fax-over-IP.

Summary: SIPStation is essentially the "official" SIP trunk for FreePBX/Asterisk, offering seamless integration and a simple pricing model. It excels in environments where ease of setup and direct FreePBX support are priorities. Compared to ClearlyIP, SIPStation is similar in many respects (flat-rate trunks, Asterisk integration) – ClearlyIP edges ahead in flexibility (with both flat and metered options) and perhaps in personalized support, whereas SIPStation benefits from Sangoma's larger infrastructure and ecosystem. For many Asterisk users, **SIPStation provides a worry-free experience**: you can enable it from your PBX GUI and trust that you're using a service built and backed by the Asterisk maintainers.

Flowroute – SIP Trunking for High-Quality Voice

Flowroute is a longstanding name in the VoIP industry and has been popular among the Asterisk community for years. Known as one of the first "pure SIP" carriers, Flowroute (now a part of BCM One) offers highly interoperable SIP trunks that work seamlessly with Asterisk flowroute.com. Many Asterisk users choose Flowroute for its call quality and flexible routing.

Asterisk Interoperability: Flowroute emphasizes adherence to SIP standards, which is why their service is *"highly compatible with Asterisk-based voice systems"* flowroute.com. They provide detailed guides for configuring Asterisk and FreePBX with Flowroute (e.g., setting up inbound routes) flowroute.com. Unlike some ITSPs that might have quirky SIP implementations, Flowroute sticks to RFC-compliant SIP, minimizing interoperability issues. A community member noted, *"I never met a SIP provider I can't connect [to] using Asterisk"*, and Flowroute exemplifies this by avoiding proprietary requirements community.asterisk.orgcommunity.asterisk.org. Flowroute supports both registration and IP authentication. With registration, you get a global format username and a choice of server. With IP authentication, you can send calls from your fixed IP directly (Flowroute was one of the early providers to offer easy IP whitelisting). Flowroute also allows multiple sub-trunks, which is useful if you have multiple Asterisk servers or want to segment traffic.

SIP Trunking Features: Flowroute offers **on-demand scalability** – there are *no fixed channel limits*, and you can burst as many simultaneous calls as your bandwidth and account balance allow flowroute.com. They charge purely per minute (no "call path" fees) flowroute.com, which we'll detail under pricing. Flowroute's network is directly interconnected with PSTN carriers; they often tout being the **"first pure SIP nationwide carrier"** with direct PSTN access flowroute.com. This means fewer hops and potentially lower latency for calls. Advanced features include support for custom caller ID (using P-Asserted-Identity header) so you can present any verified DID you own flowroute.com, as well as passing Diversion headers which is important for call forwarding scenarios flowroute.com. Flowroute also provides an **X-Tag feature** to tag CDRs for tracking calls by campaign or customer flowroute.com – a unique feature useful in call center environments.

For **failover and redundancy**, Flowroute allows customers to specify a primary PoP (Point of Presence) and will route calls via alternate PoPs if needed community.freepbx.org. They have multiple SIP endpoints (historically in US West, US East, etc.), and you can choose the closest or best-performing one. In the past, one would configure multiple `sip.flowroute.com` DNS records – now the portal allows setting a preferred PoP. Flowroute also supports inbound call failover to a secondary IP or to voicemail if your Asterisk is unreachable (configurable in their portal). On security, **TLS encryption for SIP signaling is supported** – Flowroute announced TLS support and provides instructions for using it flowroute.com. A FreePBX user confirms using TLS successfully with Flowroute (with TLS v1.2 required) community.freepbx.org. Flowroute's knowledge base also covers SIP message authentication and discusses SRTP (although it's not explicitly stated if Flowroute can handle SRTP end-to-end, many users implement SRTP at their PBX and trust Flowroute's network internally).

Supported Codecs: Flowroute supports the mainstream codecs for PSTN calls. By default, they support G.711 (PCMU/u-law and PCMA/a-law) and G.729. Many Asterisk users run G.729 with Flowroute to save bandwidth; as long as you have the G.729 codec license on Asterisk, Flowroute will pass it through. Wideband codec G.722 is accepted on the SIP trunk but note that any call leaving to the PSTN will be limited to G.711 quality. Flowroute's documentation doesn't list Opus as a supported codec, and likely Opus is not supported for PSTN trunking on Flowroute at this time (Opus would be unnecessary unless doing WebRTC-to-WebRTC calls through them, which is not Flowroute's focus). Essentially, **G.711** is the gold-standard with Flowroute (for highest quality), and **G.729** is available for low-bandwidth scenarios. Faxing is supported via T.38 passthrough on Flowroute; many Asterisk folks use Flowroute for reliable T.38 fax termination.

Pricing Model: Flowroute uses a pure **pay-as-you-go** pricing model with very competitive rates. There are *no monthly fees for channels or minimum usage requirements* flowroute.comflowroute.com. You simply pay per minute of calling and per phone number. As of 2025, Flowroute's standard rates are approximately: **$0.00833 per minute for outbound US calls** (about 0.83¢) and **$0.005 per minute for inbound** on local numbers flowroute.com. This is extremely cost-effective – for example, a 1-hour outbound call costs about $0.50. All billing is in 6-second increments, so short calls are very cheap wiki.voip.ms. Phone numbers (DIDs) cost $0.50 per month each flowroute.com, which is a flat fee. Toll-free numbers have slightly different rates (around $0.00975 per minute inbound for US/Canada toll-free flowroute.com). International calling rates vary by country; Flowroute provides a downloadable rate deck flowroute.com. They tend to have good international rates for major destinations, but as a user on Reddit discovered, some off-net destinations (like rural areas or certain countries) can be pricey if not noticed reddit.com. It's always wise to check the rate table for any country you call often.

One advantage: Flowroute doesn't charge for **concurrent calls** – you can run 100 calls at once if your account balance (prepaid) supports it. This makes it scalable for call centers or bursting traffic (whereas flat-rate trunks would require paying for 100 channels even if rarely all used). Flowroute requires maintaining a prepaid balance (refillable via credit card or ACH), and they also support postpaid billing for enterprise accounts. The bottom line: **Flowroute offers one of the most transparent and cost-efficient pricing structures** in the industry flowroute.comflowroute.com. Businesses only pay for what they use, which can yield big savings especially if call volumes fluctuate.

Security and Compliance: As a certified carrier, Flowroute complies with all FCC and CRTC regulations in North America. They were an early adopter of STIR/SHAKEN as well – being a CLEC, they sign calls originated on their network to provide full attestation. They support outbound CNAM and inbound CNAM (Caller Name) dips for a small fee per lookup if desired. For privacy and data protection, any CDR data and account info are protected under their privacy policy; EU clients may route via partners for GDPR compliance, but Flowroute primarily serves North America. On the security front, Flowroute encourages best practices like SIP credentials with strong passwords or IP whitelisting. They also maintain a **free support knowledge library** on securing Asterisk trunks. With TLS support flowroute.com, signaling security is enhanced; voice encryption via SRTP can be used from Asterisk to Flowroute's edge (though once a call hits PSTN it's out of scope).

Reliability: Flowroute has built a reputation for high reliability and call quality. They connect directly to top-tier carriers and as mentioned, act as a nationwide carrier themselves. Calls take "the most stable path possible" in and out of Asterisk flowroute.com. Flowroute's platform is geographically redundant. They have multiple edge proxy servers (historically in Seattle, Dallas, Virginia, etc.) and if one region has an issue, you can manually or automatically fail over to another. In their portal, you set a primary POP for registrations and can configure failover IP addresses for inbound calls community.freepbx.org. Asterisk servers can also register to multiple Flowroute proxies for redundancy. Flowroute claims **99.999% uptime** in some of their marketing (five-nines), though the standard SLA provided might be four-nines – specific SLA details might be given to enterprise customers. On support side of reliability, Flowroute's engineers are known to have deep SIP knowledge. They advertise that their support engineers *"troubleshoot beyond our platform and examine your entire communications system"*, with a 96% satisfaction rating flowroute.com. Phone support is available 13 hours on business days, and 24x7 coverage via email/ticket flowroute.com. This means if an issue arises in the middle of the night, you likely will use email support and get a response perhaps within a few hours. Community feedback on Flowroute is generally positive, e.g. *"Flowroute is great"* said one Asterisk user succinctly reddit.com. Another benefit: Flowroute has been around since about 2007, so they have a proven track record over many years, which inspires confidence for mission-critical use.

Support: Flowroute offers **free, un-tiered support** to all customers flowroute.com. This is somewhat unique as many providers charge for premium support tiers. You can open tickets via their online portal or by email, and their team will assist. The support hours for live help (phone or chat) cover the business day across US time zones, roughly. According to Flowroute, the fastest way to reach them is by opening a ticket or emailing, and they emphasize that no support contract is needed flowroute.com. Users have reported that Flowroute support is knowledgeable. However, there have been some complaints on forums about difficulty reaching phone support at times reddit.com – for instance, during a major outage, calls went to voicemail. These incidents appear to be exceptions, not the norm. Flowroute also provides extensive self-service documentation and an API for those who want to automate tasks (like provisioning DIDs or querying CDRs). For Asterisk users, the documentation includes step-by-step trunk configuration guides and troubleshooting for common issues (like one-way audio or NAT problems). Overall, Flowroute's support is considered **above average for technical depth**, aligning well with the needs of Asterisk professionals.

Global Reach: Flowroute's strength is in the United States and Canada. They can supply US local numbers nationwide and Canadian numbers, with SMS/MMS capabilities on many of them. They also offer international DIDs in select countries (the coverage isn't as extensive as Twilio or Telnyx – typically a few dozen countries). If your business needs numbers in, say, Europe, Flowroute might have UK and some European numbers available, but for more obscure locales you might need a secondary provider. For outbound calling, Flowroute can terminate calls worldwide; they have competitive international rates to common countries. Because Flowroute's physical infrastructure is mostly US-based, international calls from distant offices might have to traverse more latency to reach the Flowroute edge – but this is usually negligible for voice. If low-latency regional media is needed, a provider with global POPs may be preferable. In practice, many global companies use Flowroute for North America and perhaps use another trunk for other regions. It's worth noting that after Flowroute was acquired by BCM One/Intrado, they may expand global presence. But as of now, **Flowroute is best considered a North American-centric SIP trunk provider** with some international capabilities.

Summary: Flowroute is a top choice for Asterisk users who prioritize call quality, standards-compliance, and flexible pay-per-use pricing. It has a loyal following in the VoIP community, often being recommended for its reliability and support. For example, on Reddit one user wrote, *"Flowroute is great"* when asked about best SIP providers reddit.com, and an **Asterisk case study** quote praises their switch: *"we couldn't be happier with the switch to Flowroute!"* flowroute.com. The trade-offs are that Flowroute doesn't have the glitzy add-ons of Twilio (like detailed analytics dashboards or API-driven call control out of the box), and its global footprint is narrower. But for pure SIP trunking to an Asterisk PBX, Flowroute delivers outstanding performance and value flowroute.comflowroute.com.

VoIP.ms – Affordable and Feature-Rich DIY Trunking

VoIP.ms is a Canadian-based VoIP provider widely used by hobbyists, small businesses, and even some larger deployments. It's well-known in the Asterisk community for its **low costs and extensive self-service features**. VoIP.ms might not have the same enterprise brand recognition as others, but its popularity on forums (often touted as a favorite for homelab or small setups reddit.com) speaks to its reliability and value.

Asterisk Interoperability: VoIP.ms is fully compatible with Asterisk and FreePBX. They provide wiki guides for configuring Asterisk SIP or PJSIP trunks to their servers wiki.voip.ms. Users frequently mention how simple it is to set up: *"VoIP.ms works well, cheap rates and simple to setup, they even have guides for Asterisk"* reddit.com. Indeed, the provider supports both SIP registration (most common) and also allows IAX2 connections (unique among many providers) which can be useful in Asterisk-to-Asterisk trunking. Because VoIP.ms caters to a tech-savvy user base, they expose many options in their portal that integrators can tweak: e.g., you can enable/disable SIP UDP vs TCP, choose RTP encryption, select which POP (server) your trunk registers to, etc. There is no special integration module for FreePBX, but adding a trunk is straightforward. They also have an **API** which advanced users can use to provision DIDs or check balances programmatically.

SIP Trunking Features: One of VoIP.ms's strengths is the **rich feature set in their portal**. Aside from basic call routing, they offer hosted features like IVR, ring groups, voicemail boxes, call recording, and more – which you can use in conjunction with your Asterisk or even independently. For trunking specifically, VoIP.ms allows **registration to multiple servers (POPs)** but note that each sub-account (device) is tied to a single POP at a time. They have *dozens of POPs* in different cities: for example, multiple in the US (New York, Atlanta, Chicago, Los Angeles, etc.), in Canada (Montreal, Toronto, Vancouver), and some in Europe (London, Amsterdam) and elsewhere wiki.voip.ms. You choose the server closest to your Asterisk for best latency wiki.voip.ms. However, historically each DID is homed on a particular POP – meaning if that POP goes down, inbound calls to that DID fail unless you manually switch routing (VoIP.ms was working on a clustered POP architecture to improve this voip.ms). Recently, they launched a **POP clustering beta** to provide redundancy, indicating they are improving reliability by not binding a DID to a single server voip.ms.

VoIP.ms supports **failover settings**: for each DID, you can specify failover destinations (e.g., if your Asterisk is unreachable, send the call to another VoIP.ms server, to voicemail, or to a PSTN number). They also support **multiple codecs and encryption** on their trunks. A notable advanced feature: VoIP.ms now offers **TLS and SRTP encryption**. In the account settings you can enable "Encrypted SIP Traffic" which allows your sub-account to use TLS/SRTP wiki.voip.ms. Their wiki has guides for FreePBX and Asterisk on how to configure TLS/SRTP to VoIP.ms wiki.voip.ms. This is a relatively new feature (last few years). Another service they provide is **SMS** on DIDs in the US/Canada – you can send/receive SMS through their API or portal (and some have integrated it to send SMS via Asterisk).

In terms of call handling, VoIP.ms gives fine-grained control: you can set per-DID inbound routes (to your trunk, or to their digital receptionist, etc.), and for outbound, you can choose "value" vs "premium" routing per call or per account to balance cost vs quality wiki.voip.ms. They also support T.38 faxing on many routes. Features like **Caller ID filtering**, **telemarketer block (telemarketing screening)**, and **failover to secondary trunks** make it quite feature-rich. Essentially, VoIP.ms provides a toolbox that an integrator can leverage on top of the raw SIP trunk – it's a DIY paradise for those who like control.

Supported Codecs: VoIP.ms supports the following audio codecs on their trunks: **G.711u (PCMU)**, **G.711a (PCMA)**, **G.729a**, **GSM**, and as of recently **G.722** community.voip.mscommunity.voip.ms. Initially, their service officially listed only G.711u, G.729a, and GSM as supported, but they added G.722 wideband ("HD voice") capability for calls between HD endpoints community.voip.mscommunity.voip.ms. By default, G.722 is not enabled on sub-accounts (possibly to avoid transcoding issues), but users can enable it. The community forum noted G.722 is a "recent addition" and wondered why it's not default; a VoIP.ms rep explained that while G.722 provides much better audio, it uses the same bandwidth as G.711 (64 kbps) and a bit more processing, plus calls to PSTN will still be narrowband community.voip.ms. As such, they leave it optional. **Opus** is *not* supported by VoIP.ms at this time – typical for a provider focusing on PSTN termination. If you try to negotiate Opus, their servers will simply not include it in the SDP. For most Asterisk setups, this is fine since G.711 covers standard needs. The presence of GSM codec support is useful for very low bandwidth (13 kbps) scenarios or legacy devices. G.729a is fully supported; many customers use it to make multiple calls over limited internet (VoIP.ms even sells G.729 licenses if needed). Summarily, VoIP.ms covers all commonly used codecs that Asterisk supports (except Opus), giving flexibility in audio quality vs bandwidth trade-offs.

Pricing Model: VoIP.ms is known for its **ultra-competitive pay-as-you-go pricing**. Everything is à la carte, and you pay only for what you use. Key elements of their pricing:

- **DID (Number) Pricing:** You have two choices for most US/Canada local numbers – *per-minute plan* or *flat-rate plan*. The **per-minute plan** costs about $0.85/month for the DID and $0.009 per minute for incoming calls wiki.voip.ms. There's also a one-time setup fee of $0.40 for new numbers wiki.voip.ms. Under this plan, if you receive, say, 1000 minutes of calls in a month, you'd pay $0.85 + (1000 * $0.009) = $9.85 for that number. The **flat-rate plan** costs about $4.25/month for the DID with *no per-minute charge for incoming*, up to 3500 incoming minutes wiki.voip.ms. However, the flat-rate plan allows only 2 simultaneous calls on that DID wiki.voip.ms (sufficient for many small businesses). This is intended for residential or small office use where call volume is modest but predictable. It's nice that VoIP.ms offers both – you can choose flat-rate for a main number if you expect lots of inbound, or per-minute if you have many numbers or low inbound volume. Outgoing minutes are charged regardless of DID plan.

- **Outgoing Call Rates:** For the US 48 states, **$0.01 per minute** via the premium route wiki.voip.ms. They also have a **value route** for Canada (and some international) which is cheaper: e.g., Canada value is $0.0052 (half a cent) per minute wiki.voip.ms. For US calls, essentially $0.01 is the rate (they may have a slightly cheaper value route to certain US destinations, but the difference is minimal and usually they use premium for US). Canada calls can be as low as $0.0052 on value or $0.009 on premium wiki.voip.ms. International rates vary widely, but generally quite good—VoIP.ms often uses wholesale routes and passes savings on. They do, however, differentiate quality: the *premium route* uses Tier-1 carriers and guarantees CLI delivery, DTMF, etc., while the *value route* might use lower-cost carriers that could occasionally have issues wiki.voip.ms. The user can choose on each call which to use (or set a default in the portal). Most business users stick to Premium for critical calls (still only 1¢/min in US) wiki.voip.ms. Inbound calls on toll-free numbers are $0.019/min in the US wiki.voip.ms, which is standard, and toll-free DIDs cost $0.99/month wiki.voip.ms.

- **Miscellaneous:** VoIP.ms bills in 6-second increments for almost all calls wiki.voip.ms, which yields slight savings over per-minute rounding. They have no monthly minimum—one can spend just $1 a month if that's all the usage. This makes it extremely attractive for low-volume scenarios (home, small office, or backup trunks). Services like E911 cost $1.50/month per location wiki.voip.ms, and CNAM dip (caller name lookup) is $0.008 per lookup wiki.voip.ms. These are optional add-ons. Essentially, **VoIP.ms is one of the most cost-effective providers**, often cited by users: *"Put $15 in and you're good for a year"* in a homelab context reddit.com.

Security and Compliance: VoIP.ms now offers robust security options. With TLS/SRTP available wiki.voip.ms, one can secure their SIP signaling and audio. Each sub-account (which corresponds to a trunk registration) can be locked down to specific IP addresses as well, providing IP-based security. They support secure password policies and two-factor authentication for the portal. Since they operate in Canada and serve many U.S. customers, they adhere to privacy laws and likely can accommodate GDPR concerns (they have servers in the EU). They are fully compliant with **STIR/SHAKEN** for U.S. calls – in fact, by using Tier-1 termination carriers for premium route, they ensure caller ID integrity wiki.voip.ms. For HIPAA, VoIP.ms doesn't explicitly advertise compliance or signing BAAs (as they target mostly SMB, not healthcare specifically), so cautious healthcare clients might use a more enterprise-oriented provider. Nonetheless, the voice data on their network can be encrypted and they have secure datacenters. They have an active fraud monitoring team; for example, accounts have daily spending limits by default and can be configured with IP restrictions, to prevent a compromised Asterisk from running up a huge bill – an important consideration.

Reliability: VoIP.ms has a generally good reliability record, though not without some past hiccups. Because of their architecture with many independent POPs, occasionally a single POP might have an issue (meaning customers registered to that server experience problems). The recommended practice is to have a script or monitoring that can switch your trunk to another POP if one goes down, or use their failover settings. The introduction of **clustered POPs** in 2022/2023 aims to mitigate this by pooling resources across locations voip.ms. They have also diversified their carrier partners to improve call completion. In 2021, VoIP.ms suffered a notable DDoS attack that impacted service for several days – a rare but significant event, after which they invested in stronger DDoS protection and network resilience. In normal times, their uptime is quite solid. The service doesn't come with a formal SLA for refunds unless you're a large customer, but they often compensate if there's a major outage. Voice quality using premium routes is excellent; using Tier-1 carriers means calls are clear and stable wiki.voip.ms. Asterisk users often run multiple trunks (e.g., VoIP.ms as primary and another as backup) to ensure redundancy – a wise strategy with any single provider, including VoIP.ms.

Support: VoIP.ms provides support primarily through an **email/ticket system and a community forum**. They do not offer phone support for regular accounts (this keeps costs low). However, their support team is quite responsive on tickets – typically responding within a couple of hours during business hours, and they do have 24/7 staff for network operations. The community forum (and a Discord channel unofficially) has VoIP.ms staff and knowledgeable users who help each other. For configuration help with Asterisk, their wiki is detailed and many tutorials exist. Because VoIP.ms is popular among DIYers, one can find a lot of third-party blogs and forum posts about integrating it with Asterisk, which helps new users. Reviews often praise VoIP.ms for giving "the best bang for buck" and having a stable service, while acknowledging that it's mostly self-service. In summary, support is adequate for experienced users (with lots of documentation and community help), but enterprise users who prefer phone support might find it less direct than providers like Twilio or Flowroute.

Global Reach: VoIP.ms has a decent global reach in terms of DIDs and servers. They offer **local numbers in over 60 countries**, including most of North America, much of Europe, and parts of Asia and Latin America. This allows businesses to have international presence. The rates for international DIDs vary by country, and some are only offered on a per-minute incoming basis. The multiple POPs in different continents mean you can register an Asterisk server in, say, London to their London server for local traffic. However, one limitation historically was that a DID from one country might only route to certain servers. The ongoing improvements in their architecture aim to make it more cloud-like and less tied to individual servers. For outbound, VoIP.ms can complete calls worldwide as well. The value vs premium distinction applies internationally too – e.g., you might get a slightly cheaper route to some country if you don't mind variable quality. Businesses with heavy global needs might use VoIP.ms in combination with another global provider to ensure optimal routing, but many find VoIP.ms sufficient for international calls at low volumes. They don't have as many POPs in Asia/Australia (none in Australia, one in Singapore in beta, etc.), so latency for those regions could be higher. Still, for a provider of its size, VoIP.ms's global offerings are impressive.

Summary: VoIP.ms is often **recommended for Asterisk home labs and small businesses** due to its low cost and rich features reddit.com. It brings a lot of value – essentially acting as both a SIP trunk provider and a mini-hosted PBX if you want it to. Its interoperability with Asterisk is proven and straightforward reddit.com. The trade-off for the low price is that you must be comfortable with a DIY approach: managing your own failover, doing your own configuration, and using online support channels. For many, that's perfectly fine and even empowering. In comparisons, VoIP.ms might not match the ultra-high SLAs or global network of Telnyx/Twilio, but it punches above its weight with features and often becomes a **favorite for cost-conscious deployments**.

Telnyx – Global High-Performance SIP Trunking

Telnyx is a newer entrant (founded mid-2010s) that has quickly become a top-tier SIP trunk provider with a strong appeal to both developers and telecom professionals. Telnyx offers **"Elastic SIP Trunking"** over a private global IP network, combining cloud-like scalability with traditional telecom reliability. It is frequently compared to Twilio, but often at lower cost and with more telecom-focused features. For Asterisk users, Telnyx provides extensive documentation and an advanced feature set.

Asterisk Interoperability: Telnyx is fully compatible with Asterisk and FreePBX. They publish configuration guides for Asterisk, FreePBX, 3CX, and many other platforms support.telnyx.comsupport.telnyx.com. Setting up Telnyx with Asterisk typically involves creating a SIP Connection in the Telnyx portal (with credentials or IP auth) and then configuring a PJSIP trunk on Asterisk with those details. Notably, Telnyx supports both UDP and TLS transports for SIP, and registration or IP-based authentication. Asterisk users have reported easy setups; one Reddit user said simply, *"Telnyx.com works"* when discussing trunks reddit.com. Telnyx's adherence to SIP standards is strong, and they even support newer features like SIP URI dialing and video calls through SIP (H.264 codec, etc.) support.telnyx.com. The company's developer-friendly ethos means they provide APIs and tools that can be integrated with Asterisk if you want to do things like dynamic call control via REST API in parallel with your PBX. But using Telnyx purely as a standard SIP trunk is straightforward and well-supported.

SIP Trunking Features: Telnyx offers an impressive range of features in their SIP trunking product:

- **Elastic Scaling:** There are no fixed channels – capacity is limited only by your network and account settings. Telnyx can handle burst traffic and high CPS (calls per second) for dialer applications smoothly.
- **Global Infrastructure:** Telnyx operates its own **private global IP network** with points of presence around the world. They have **17 PoPs in high-density metros** across North America, Europe, Asia, and Oceania telnyx.com. Traffic from your Asterisk can connect to the nearest Telnyx edge, reducing latency. Their network uses anycast and private fiber interconnects for reliability and quality telnyx.com.
- **Failover and High Availability:** You can register your Asterisk to multiple Telnyx SIP proxies (they provide regional SIP domains, e.g., sip.telnyx.com resolves to multiple IPs). If one path fails, calls can failover automatically. They also support active-active registration. Additionally, you can configure failover routing on their portal: e.g., if your PBX is unreachable, Telnyx can forward calls to another trunk or number.
- **Encryption:** Telnyx fully supports **TLS 1.2/1.3 for SIP signaling and SRTP for media** sip.telnyx.com. This is a first-class feature, not an afterthought, meaning you can enable "Secure Trunking" in their portal easily. This appeals to security-conscious deployments.
- **Codec Support:** Telnyx has one of the broadest codec supports. They support all the common ones and more: G.711 (u-law/a-law), G.729, G.722, GSM, and even **Opus** support.telnyx.com. In their FreePBX configuration, they explicitly instruct enabling Opus along with traditional codecs support.telnyx.com, which shows Telnyx can negotiate Opus on the SIP trunk. This is useful if you have WebRTC clients or want end-to-end HD voice (Opus to Opus) between two VoIP endpoints via Telnyx. They also support T.38 for fax and even video codecs (H.264) for video calling scenarios support.telnyx.com.

- **Advanced Features:** Telnyx provides a lot of extras: They have an **API** that can control call routing and retrieve CDRs in real-time, which means you can integrate Telnyx with your applications (though that's more relevant for Programmable Voice use-cases). For trunking, noteworthy features include **Caller ID name (CNAM) storage** – you can register a CNAM for your numbers, **E911** provisioning with address verification, and **global outbound CLI** (they can send your caller ID internationally where allowed). Their portal also allows configuring SIP header manipulation, if needed, and setting up multiple connection profiles for different traffic types (useful if you segregate traffic).
- **SMS and MMS:** Telnyx DIDs in many countries support SMS/MMS. While not directly an Asterisk trunking feature, it's part of their offering if you want to integrate messaging (some have connected Telnyx SMS with FreePBX using custom scripts or their API).

Supported Codecs: As mentioned, Telnyx supports all major voice codecs used in VoIP. Specifically, **G.711 μ-law and A-law** are supported (and typically the default for PSTN calls), **G.729** is supported for low-bandwidth needs support.telnyx.com, **G.722 and Opus** are supported for high-fidelity voice (Opus is often used if you have WebRTC clients or between two VoIP endpoints on the Telnyx network) support.telnyx.com. They also support legacy codecs like GSM and iLBC, although those are less commonly used now. Telnyx's documentation suggests enabling ulaw, alaw, gsm, g722, g729, and Opus – which they label as **"Telnyx-supported codecs"** support.telnyx.com. This wide support means you can confidently use modern codecs. For example, if you have an Asterisk-based call center with agents on WebRTC softphones, you could run Opus between agent and Asterisk, then Asterisk-to-Telnyx also in Opus, and only transcode to G.711 at Telnyx's PSTN gateway (or let Telnyx handle transcoding). This could improve audio quality for calls that stay on-net and reduce bandwidth. On fax, Telnyx supports T.38 and G.711 passthrough – reliably handling faxes is part of their telecom focus. In summary, **Telnyx's codec flexibility is state-of-the-art**, surpassing many competitors (for instance, Twilio only recently added Opus and still treats it as limited availability twilio.com).

Pricing Model: Telnyx uses a **pay-as-you-go pricing** model with very competitive rates, often lower than Twilio for similar quality. They have *no monthly trunk fees or channel fees*, and you can buy numbers and minutes à la carte. Some highlights of Telnyx pricing:

- **Outbound Calls (Termination):** Starting around $0.007/min for US calls twilio.com. This can go lower with volume discounts. They have automatic volume tiering or you can commit to a monthly spend for better rates telnyx.comtelnyx.com. For example, large volumes might drive the rate under $0.005/min. International termination rates are country-specific but generally on par with or slightly better than Flowroute/VoIP.ms for many destinations.
- **Inbound Calls (Origination):** $0.001 or $0.002 per minute for inbound US calls (Telnyx's site indicates a small per-minute fee on inbound, unlike some that give free inbound). However, they often offset this by low monthly DID costs or bundles. In practice, inbound might effectively be very low cost.
- **DIDs:** US local numbers cost about **$1.00 per month** telnyx.com. They also offer "bundled" channels – for instance, they have an option where for $0.99/month extra you get unlimited inbound on that DID (like a virtual channel). Toll-free numbers are a bit more (around $2). International numbers vary but Telnyx has good coverage (50+ countries) and prices fairly.
- **Volume/Contract Discounts:** Telnyx offers **contract pricing** that gives you predictable bills and lower rates at high volumes telnyx.comtelnyx.com. They mention that if you get on a contract, you benefit from *24/7 support and a dedicated customer success manager* telnyx.com, implying that larger customers get premium service at no extra charge beyond the commitment.
- **Billing increments:** Typically 60/60 for international, 6-second for domestic (this detail would be in their specs; many US trunks do 6-second).
- **Comparison:** For a sense of cost, Telnyx can be 20–30% cheaper than Twilio on pure per-minute and per-DID fees, which is significant over large call volumes.

No flat-rate unlimited channels here – Telnyx's philosophy is usage-based, which often ends up more economical unless you constantly max out all channels (in which case a flat might benefit you). The pay-for-what-you-use model plus *no contracts required* (for standard pricing) means you can start small and scale up without needing to reconfigure anything.

Security and Compliance: Telnyx takes security seriously. With support for **TLS 1.3 and SRTP** sip.telnyx.com, they ensure call data can be encrypted end-to-end. They have guides and a support team to help implement secure trunks. On regulatory compliance, Telnyx, being a global provider, adheres to GDPR for EU user data, can sign BAAs for HIPAA (they have case studies with healthcare), and participates in U.S. telecom compliance like STIR/SHAKEN. In fact, Telnyx has been actively issuing calls with full attestation under STIR/SHAKEN and offers tools to help customers ensure their calls aren't flagged (they have a portal to manage call signing status). For 911, Telnyx provides E911 service in the US and equivalent emergency access in other countries where applicable, with an easy interface to assign emergency addresses to DIDs. They also support **call encryption and anonymization features** for privacy – for example, you can use their API to mask phone numbers (similar to Uber's use case). Internally, Telnyx's network security is robust: by spanning across AWS, GCP, and Azure clouds with private links telnyx.com, they mitigate many single points of failure and maintain security through network isolation. They also have DDoS protection in place given their global network.

Reliability: Telnyx is often praised for its reliability and performance. Their architecture is multi-cloud and geographically distributed, which provides strong resilience against outages telnyx.com. If one cloud provider has an issue, Telnyx can route traffic through another. With **17 PoPs worldwide** telnyx.com, calls are kept on-net as long as possible, reducing dependency on the public internet. Telnyx claims very high uptime (they likely target 99.99% or better). They publish status on their status page and have a NOC monitoring 24/7. Because they control their network, jitter and packet loss are minimized – this results in consistently clear calls. Some users have reported that Telnyx call quality and latency are among the best, sometimes even better than Twilio's, likely due to their network optimizations.

Support: Telnyx offers **24/7 support** for all customers via chat/email, and phone support for critical issues especially if you have a contract telnyx.com. They pride themselves on being developer-friendly and responsive. Many users in forums note that Telnyx's live chat support is quick and knowledgeable (sometimes even helping with Asterisk SIP settings). For larger customers, Telnyx assigns a customer success manager, and they also have solution engineers who can assist with onboarding (for example, configuring your trunking or porting numbers). Their documentation is extensive – including a detailed support center with articles on TLS/SRTP support.telnyx.com, hardware configuration, number porting, etc. If you compare with Twilio, one often hears that Twilio's support can be slow unless you have a paid plan, whereas Telnyx tends to be more accessible at base level. On the FreePBX forums and Reddit, Telnyx is frequently recommended for their support. As an anecdote: one of the Reddit users in **/r/Asterisk** identified themselves as a Telnyx employee when helping someone, showing their engagement with the community reddit.com.

Global Reach: Telnyx truly shines in global reach. They have a **private backbone connecting North America, South America, Europe, Asia, and Australia**. They offer local numbers in dozens of countries (from common ones like UK, Germany, Australia, to more niche ones). They have local media anchors – meaning if you're making a call from Europe via Telnyx, the RTP can stay in Europe through a local Telnyx gateway, improving call quality. This is in contrast to many providers (like VoIP.ms or Flowroute) where all media might trombone back to the US. Telnyx's global coverage also extends to emergency services; e.g., they provide emergency calling in EU countries (with proper address registration). For multinational businesses or Asterisk deployments in various regions, Telnyx is a great option because you can manage all trunks in one portal and get consistent features worldwide. They also support **multi-regional registration** – you can have your Asterisk register to the closest regional domain (e.g., `sip.eu.telnyx.com` for Europe, `sip.sydney.telnyx.com` for Australia, etc.), but still manage everything in one account. This is ideal for latency and compliance (calls stay within region when needed).

Summary: Telnyx combines the best of two worlds – the **telecom reliability** of a traditional carrier and the **flexibility and innovation** of a tech company. For Asterisk users, Telnyx offers an excellent mix of features: high-quality global call routing, modern security, wide codec support (including Opus) support.telnyx.com, and competitive pricing. It's no surprise that Telnyx often comes up as a recommended provider for new installations where global reach or advanced features are required (e.g., in one thread an AWS engineer recommended Amazon's own service, but another user simply stated "Telnyx works" as a go-to solution reddit.com). Telnyx can be seen as a direct competitor to Twilio's Elastic SIP Trunking, generally at a lower cost and with more hands-on support. For those who need a **scalable, international SIP trunk for Asterisk**, Telnyx is a top contender, if not the leader, in 2025.

Twilio – Elastic SIP Trunking with Cloud Ecosystem

Twilio is a well-known cloud communications platform, and its **Elastic SIP Trunking** service provides global SIP trunk connectivity with the benefit of Twilio's powerful platform and tools. Twilio's offering is often attractive to larger enterprises and developers who may want to blend trunking with programmable voice/sms features. While Twilio can be more expensive, it brings proven reliability at massive scale (used by companies like Lyft, Airbnb, etc.).

Asterisk Interoperability: Twilio's SIP trunks are standards-compliant and work with Asterisk, although Twilio is not specifically focused on Asterisk in the way others are. You won't find official Twilio guides for Asterisk on their site (they cater to a broader audience), but the configuration is straightforward: you create a Twilio Elastic SIP Trunk in their console, configure your Asterisk server's IP or credentials, and define the termination URI (your Asterisk's public IP or domain). Many have successfully integrated FreePBX with Twilio; the main caveat is that Twilio prefers IP authentication for trunks (they expect you to have a static IP and whitelist it in their portal for incoming calls). Twilio does support SIP registration now in limited form but generally expects IP-based setups. There are community how-tos for using Twilio with Asterisk. Twilio's trunking uses standard SIP over UDP or TLS (they provide a list of regional SIP gateway addresses to use). One thing to note: Twilio requires all traffic to use their E.164 formatting (i.e., + dialing), which is easily handled by Asterisk dialplan rules.

SIP Trunking Features: Twilio Elastic SIP Trunking is designed to be **scalable and resilient**. Some key features:

- **Unlimited Capacity:** No channel limits – Twilio automatically scales up and down based on traffic twilio.com.
- **Multi-Region Presence:** Twilio has trunking gateways in various regions (North America, Europe, Asia-Pacific, Latin America). You can choose nearest regions for ingress/egress of calls, and also set up redundant Origination URIs for failover across data centers. Twilio can intelligently route outbound calls through the nearest media PoP to the destination for better quality.

- **Failover Settings:** You can configure primary, secondary, tertiary targets for incoming calls if one is unreachable. Twilio will attempt in order – for example, first your Asterisk, then a backup PBX or voicemail.
- **TLS/SRTP Encryption:** Twilio supports secure trunking – in fact, they highlight *"carrier-grade security"* with TLS/SRTP to protect signaling and media research.com. Enabling Secure Trunking in Twilio is as simple as toggling a setting, and they provide guides for configuring TLS on common SBCs twilio.com.
- **Programmability:** Since Twilio's trunking is part of their larger platform, you can mix in Twilio's API capabilities. For instance, you could have incoming calls hit Twilio, use a TwiML application to screen or gather info, then forward into Asterisk, or send a call from Asterisk into Twilio for IVR then back. This hybrid use is optional but powerful.
- **Number Management:** Twilio offers **geographic numbers in over 100 countries**. Managing DIDs (buying, porting) is done via their console or API. They also offer **on-demand number provisioning** via API, which can be dynamic (but for trunks, usually you statically assign numbers).
- **Advanced Features:** Twilio supports things like **SIP Registration (beta)** if you don't have a static IP, **SIP URI** domain hosting (you can have a Twilio SIP address domain and have phones register to Twilio which then trunks to your Asterisk), and **international emergency calling** in supported countries. Twilio can also do **media forking** (via SIPREC, for call recording compliance) research.com. They have an add-on for **CNAM inbound** and tools for **spam call detection (SHIELD)** which might be relevant if you want Twilio to help filter robocalls before they hit Asterisk.

Supported Codecs: Twilio Elastic SIP Trunking initially supported only G.711 (PCMU/A) as GA (generally available). However, Twilio now supports other codecs in "Limited Availability" upon request: specifically **G.729, Opus, and AMR-NB** twilio.com. According to Twilio's documentation, PCMU and PCMA are the default codecs for all trunks twilio.comtwilio.com. If you need G.729 or Opus, you have to enable those codecs on your trunk (they have a process via support to enable the Limited Availability codecs). Twilio recognizes that Opus can give superior quality and efficiency, so they started allowing it for trunking to accommodate WebRTC use cases callin.io. For example, a Twilio PDF guide for Cisco CUBE mentions Twilio trunk supports G.711u/a, G.729, G.722, SILK, and Opus, but currently actually negotiates G.711 by default twilio-cms-prod.s3.amazonaws.comtwilio-cms-prod.s3.amazonaws.com. In summary, expect **G.711** by default, and optionally **G.729 or Opus** if you specifically enable them. Twilio also supports T.38 passthrough for fax (they suggest using G.711 though, if possible). For most Asterisk scenarios with Twilio, leaving it at G.711 is simplest unless you have a strong reason to use G.729 (bandwidth) or Opus (WebRTC endpoints).

Pricing Model: Twilio's pricing is purely usage-based (no monthly trunk fees), but it's generally higher than competitors. For Elastic SIP Trunking in the US: **Inbound calls start at ~$0.0045/min** and **Outbound at ~$0.0070/min** twilio.com. That means incoming 0.45¢, outgoing 0.7¢ per minute. This is actually in line with Telnyx's base rates. Twilio's phone numbers are around $1.15/month for local numbers in the US mycountrymobile.com, slightly higher than others mycountrymobile.com. International rates vary; Twilio tends to have very good coverage, but sometimes at a premium cost. However, Twilio offers **volume discounts** and committed-use discounts. Large customers negotiate custom rates, which can be competitive with other providers. Twilio also does not charge for concurrency or trunk registration. One difference: Twilio bills per minute (no 6-second increment on trunks; they round up to nearest minute, which can increase cost for short calls). They often have minimum call charges of 1¢ for very short calls.

For example, a small business might find Twilio's bill a bit higher compared to VoIP.ms or Flowroute for the same usage, but Twilio's value might be seen in its reliability and the array of services. It's worth noting Twilio has a **$0.0052 per call carrier fee** on inbound toll-free (and some surcharges for certain area codes) and other telecom compliance fees that get added, which simpler providers might not break out. Twilio's transparency and detail here is good but it does mean careful reading of their pricing sheets.

Security and Compliance: Twilio is enterprise-grade in compliance. They have all relevant certifications (SOC II, ISO 27001, etc.), and they sign BAAs for healthcare. Twilio supports **TLS/SRTP** to secure calls research.com, which is crucial for many enterprise deployments, and provides guides on how to implement it twilio.com. They fully implemented **STIR/SHAKEN** as of 2021; calls through Twilio trunks are signed and Twilio will honor attestation on incoming calls. Twilio also offers features like **Secure Trunking** (where media can be delivered encrypted and directly to your premise via VPN or private connectivity using Twilio Interconnect). For global regulatory compliance: Twilio can help with things like ensuring CLI rules are followed in each country, and they will disable features where not allowed (for example, they know where local laws require a local presence to present a number, etc.). Twilio's platform also has fraud detection – e.g., they monitor traffic for anomalies and can auto-block suspicious call patterns (with alerts to the customer). Additionally, Twilio is GDPR compliant and has data residency options if needed. In short, **Twilio meets or exceeds telecom compliance standards** and provides tools to customers for security (like IP access control lists on trunks, etc.).

Reliability: Twilio is famous for its reliability. Their infrastructure spans multiple AWS regions and uses redundant infrastructure. They advertise **"carrier-grade"** service with global redundancy, and indeed many mission-critical services use Twilio. The trunking service has a financially-backed SLA (often 99.95% or 99.99% uptime depending on plan). Twilio's approach to reliability includes intelligent routing – if one carrier to a destination is down, Twilio automatically re-routes via another (they have a huge network of interconnects). They maintain voice quality by monitoring call quality metrics. One example of reliability engineering is Twilio's Super Network: it chooses the best path for each call in real-time, and can failover fast. From an Asterisk user perspective, Twilio's reliability means you rarely have to worry about the trunk being the point of failure; focus can be on your own

PBX uptime. Twilio also offers an **SLA for call completion and availability** if you opt for their Enterprise add-ons. In practice, Twilio's trunking has had extremely few outages (when compared to smaller providers). That said, if Twilio does go down, it can affect a lot of customers at once (centralized impact). Some risk mitigation is possible by having a backup provider, but many trust Twilio as sole trunk because of their track record.

Support: Twilio's support is a bit tiered. By default, standard developer accounts rely on **email support and community forums**. Response times can be one business day or more for non-critical issues. Twilio has a very active developer community and documentation, which helps self-service. For businesses, Twilio offers paid support plans (e.g., a gold or platinum support plan) which provide 24/7 phone support and faster response, but these come at additional cost unless you have a large commitment. However, Twilio's Enterprise customers (and any sizable SIP trunking customer) often get dedicated account managers and solution engineers who ensure the deployment runs smoothly. Twilio also has extensive docs for every feature (for instance, securing trunks, using their API for trunking stats, etc.). Compared to others: Twilio's baseline support may feel less personal – you're dealing with a big company. But their reliability often means less need for support. For Asterisk-specific issues, Twilio might not help debug your dialplan, but they will ensure the SIP signaling and their side is correct. The **Research.com review** of Twilio trunking notes that Twilio provides *"detailed analytics and reporting"* research.com and integration options with CRMs, etc., which can be seen as a form of support – giving users tooling to help themselves. Ultimately, Twilio's support is excellent with the right plan but could be slower at the basic level; organizations should weigh if they need hands-on assistance (in which case Telnyx or ClearlyIP might be friendlier) or if they are comfortable with Twilio's robust self-service approach.

Global Reach: Twilio has one of the **broadest global reaches** of any SIP trunking provider. They offer phone numbers in **over 100 countries** and have outbound call termination to just about every country on the planet. Twilio's network has media gateways in multiple regions (North America, South America, Europe, Asia-Pacific) so calls typically don't have to travel to the other side of the world just to exit to PSTN. The ability to have local numbers virtually anywhere is a big draw for Twilio. For example, a business could get numbers in all EU countries and manage them in one Twilio account, whereas some smaller providers might not cover each country. Twilio also handles regulatory compliance for phone numbers – if a country requires address registration or proof of residency to obtain a number, Twilio guides you through that. Their reach also extends to emergency services: Twilio supports emergency calling in 40+ countries by leveraging local carrier partnerships, which is critical for multinational deployments. Additionally, Twilio's global reach includes **local connectivity** – in some regions they have direct interconnects with telecoms (especially since Twilio acquired an international carrier (ValueFirst and others) and partners with providers like SingTel, etc.). Twilio's Super Network essentially is a patchwork of global carrier links accessible via one cloud API. For a globally distributed Asterisk deployment, Twilio can simplify management by centralizing all telephony needs. The trade-off might be cost, as Twilio could be pricier in some markets versus using a local SIP trunk provider.

Summary: Twilio Elastic SIP Trunking is a premium service that offers **reliability, global coverage, and deep integration capabilities**. It's an excellent choice when a business is already using Twilio's platform or needs worldwide numbers with minimal fuss. Twilio brings features like *secure TLS/SRTP, analytics, multi-region trunks, and contact center integration* research.comresearch.com that appeal to enterprises. A 2025 review highlights Twilio's support for *"multiple audio codecs (G.711, G.729, Opus) to balance bandwidth and quality"* and *"multi-region deployment to optimize call routing"* research.comresearch.com, confirming its versatility. For Asterisk-specific use, Twilio may require a bit more manual setup (no native FreePBX module, etc.), but it is undoubtedly used successfully in many Asterisk installations. Businesses that value Twilio's broader ecosystem (like using the same platform for SMS, WhatsApp, or building voice apps) will find Twilio trunks a natural fit. In short, Twilio sets the bar for a **cloud-based SIP trunk with global scale**, though that bar comes at a higher price and complexity that not every Asterisk deployment needs.

Other Notable Providers (Briefly)

In addition to the major providers above, there are a few others worth mentioning for Asterisk SIP trunking:

- **Bandwidth/Level 3/Inteliquent (Wholesale Carriers):** Companies like Bandwidth (which acquired Voxbone), Lumen (Level 3), and Inteliquent (now Sinch) are actual backbone carriers that offer SIP trunking, often to large enterprises or other providers. They boast extremely high reliability and extensive U.S. number inventories. These carriers require a vetting process and usually a contract. As one VoIP forum user noted, *"Sinch (Inteliquent) will vet you through the roof and by the time you're done...you know you are good to do business into the USA."* voip-info.org. For Asterisk, using a wholesale carrier can work well if you have the scale (they expect high volume) and telecom expertise to manage direct carrier interop. Otherwise, many smaller ITSPs (including ClearlyIP and others) actually resell these carriers in the backend.
- **SignalWire:** SignalWire is a newer CPaaS founded by the original creators of FreeSWITCH. They offer SIP trunking at extremely aggressive prices (their tagline includes $0.0007/min rates for some calls) reddit.com. SignalWire's network is global and focused on programmability, similar to Twilio, but they aim to undercut Twilio's pricing. For Asterisk, SignalWire can be an interesting option if you want Twilio-like API capabilities, very low rates, and don't mind a less mature platform. They have a FreePBX module for integration as well. While not as battle-tested as others, SignalWire is growing in popularity for cost-sensitive applications.

- **Amazon Chime Voice Connector (AWS):** Amazon Web Services provides a service called Voice Connector under its Chime product line, which is essentially SIP trunking hosted on AWS. It's priced around $0.003/min and is intended to integrate with on-prem PBXs (including Asterisk) or UC systems. An AWS engineer even created a sample for connecting Asterisk to Chime Voice Connector reddit.com. The service is reliable (running on AWS's infrastructure) and particularly attractive if your Asterisk is hosted in AWS. However, it's a bit barebones (no SMS, fewer features), and available mainly in US and select countries.
- **Voxbone (now Bandwidth):** Historically, Voxbone was a go-to for global DIDs and SIP trunking (with a strong European presence). They were acquired by Bandwidth. If you need a lot of European numbers and high quality, Bandwidth (via the old Voxbone network) is excellent, though generally targeted at larger enterprises with commitments.
- **Other Regional Providers:** Depending on your region, there are local providers that specialize in SIP trunks for Asterisk. For example, **Andrews & Arnold (A&A)** in the UK was mentioned by a user for good rates in the UK home lab context reddit.com. In Australia, providers like **Twilio (AU) or Maxo** might be considered. In the EU, **Deutsche Telekom** offers SIP trunks certified with Asterisk. These can sometimes integrate tightly with local telco requirements (like emergency services, local number portability). When evaluating region-specific providers, ensure they support SIP standards and ideally find references of other Asterisk users' experience with them.
- **Cloud PBX Providers Offering Trunks:** Some cloud PBX vendors (e.g., 3CX, RingCentral) sometimes allow a SIP trunk handoff to an on-prem system, but this is less common and often not officially supported. It's usually more cost-effective to use a dedicated SIP trunk provider for Asterisk rather than a PBX provider's trunk.

Each of these additional options has its niche – be it ultra-low cost, specific regional strengths, or integration with existing cloud ecosystems. Professionals should weigh these against the major players we detailed above.

Comparison of Providers

Below is a **comparison table** summarizing key attributes of the major VoIP providers for Asterisk discussed in this report. This offers a side-by-side view of interoperability, features, and other criteria:

PROVIDER	ASTERISK INTEGRATION	KEY SIP TRUNK FEATURES	SUPPORTED CODECS	PRICING MODEL	SECURITY & COMPLIANCE	REL SUP
ClearlyIP	Native FreePBX module; guides for Asterisk kb.clearlyip.comclearlyip.com.	Multi-server registration & SRV failover kb.clearlyip.com; E911 included clearlyip.com; STIR/SHAKEN enabled clearlyip.com.	G.711 (ulaw/alaw), G.729, G.722 (likely); Opus not explicit.	**Flat-rate channels** ($17.99–$24.99/channel) or **$0.009/min** metered clearlyip.comclearlyip.com; DIDs ~$1 clearlyip.com.	TLS/SRTP supported (by request); E911, Kari's Law/Ray Baum's Act compliant clearlyip.com; Calls signed with STIR/SHAKEN clearlyip.com.	Red & Ca kb.c (reg for I prai com com
Sangoma SIPStation	Built into FreePBX (one-click setup); certified with Asterisk.	Inbound call failover to backup PSTN sipstation.com; tight integration with Sangoma UC products sipstation.com.	G.711 (u/a), G.729; G.722 supported internally; no Opus.	**Flat-rate per channel** ($19.99–$24.99/mo) sipstation.comsipstation.com; no per-minute on local calls; DIDs ~$1.	TLS/SRTP capable (FreePBX config); E911 for each DID; STIR/SHAKEN compliant (per FCC).	Geo cent upti hour (tick stor Free
Flowroute	Official Asterisk guides flowroute.com; RFC-compliant SIP flowroute.com.	No channel limits flowroute.com (elastic capacity); multi-PoP routing (choose preferred POP) community.freepbx.org; advanced SIP headers (PAI, Diversion) flowroute.com.	G.711 (ulaw/alaw), G.729; G.722 unofficially (narrowband focus); no Opus.	**Pay-as-you-go**: ~$0.00833/min outbound, $0.005/min inbound flowroute.com; DIDs $0.50/mo flowroute.com; no contracts flowroute.com.	TLS for SIP signaling supported flowroute.com; SRTP not explicit; STIR/SHAKEN fully implemented; CNAM dips available.	Dire netw qual 96% supp supp flow free flow
VoIP.ms	Asterisk setup wikis reddit.com; supports SIP and IAX2.	Many self-service features (IVR, voicemail, SMS); multiple POPs to choose wiki.voip.ms; DID failover configurable.	G.711u/a, G.729a, **G.722** (HD) community.voip.ms, GSM; no Opus.	**Pay-as-you-go**: US/CA $0.01/min outbound wiki.voip.ms, $0.009/min inbound wiki.voip.ms; DIDs $0.85/mo + usage or $4.25 flat (3500 min) wiki.voip.mswiki.voip.ms; 6-sec billing.	TLS/SRTP available (encrypted trunk option) wiki.voip.ms; E911 $1.50/mo wiki.voip.ms; STIR/SHAKEN via premium routes wiki.voip.ms.	Dec (cho wiki auto failo beta via t com (res pho
Telnyx	Config guides for Asterisk/FreePBX support.telnyx.com; very SIP-standard friendly.	**Global PoPs (17)** telnyx.com with private network; active-active redundancy; SMS/MMS on DIDs; API for control.	G.711 u/a, G.729, **Opus** support.telnyx.com, G.722, GSM; H.264 video supported support.telnyx.com.	**Pay-as-you-go**: ~$0.007/min outbound twilio.com, ~$0.002/min inbound; DIDs ~$1.00 telnyx.com; volume discounts & contracts cut rates telnyx.com.	TLS v1.2/1.3 and SRTP fully supported sip.telnyx.com; HIPAA and GDPR compliant; STIR/SHAKEN on all calls; robust anti-fraud tools.	Mult (five aim) (free teln CSM teln rega tear

PROVIDER	ASTERISK INTEGRATION	KEY SIP TRUNK FEATURES	SUPPORTED CODECS	PRICING MODEL	SECURITY & COMPLIANCE	RELI SUP
Twilio	Works with Asterisk (IP auth preferred); no native module, but standard SIP.	Unlimited capacity (elastic) twilio.com; multi-region trunks; rich APIs (programmable voice); number porting in 100+ countries.	G.711 (PCMU/A) default twilio.com; **G.729, Opus** available on request research.com; supports T.38 fax.	**Pay-as-you-go**: ~$0.0070/min outbound, $0.0045/min inbound US twilio.com; DIDs ~$1.15/mo mycountrymobile.com; enterprise volume discounts.	TLS/SRTP encryption supported research.com; extensive compliance (BAA, GDPR, etc.); carrier-grade security (call encryption, fraud detection) research.com.	High glob SLA basi ema avai plan & m rese

kb.clearlyip.com clearlyip.comflowroute.com reddit.comsupport.telnyx.com research.com

Table: Comparison of leading SIP trunk providers for Asterisk across key factors. (Sources: ClearlyIP kb.clearlyip.comclearlyip.com; Flowroute flowroute.com; VoIP.ms reddit.com; Telnyx support.telnyx.com; Twilio research.com and others as cited above.)

Conclusion and Recommendations

Choosing the "best" VoIP provider for an Asterisk PBX depends on the organization's specific needs – whether it's cost optimization, advanced features, global coverage, or top-tier support. **ClearlyIP** is an excellent first choice for those in the Asterisk/FreePBX community who value seamless integration and hands-on support; it checks all the boxes for SMBs with a mix of flat-rate and metered plans and Asterisk-focused service. **Sangoma SIPStation** similarly serves Asterisk users who want a plug-and-play official solution with flat-rate pricing and reliable performance in North America. If **cost and community trust** are paramount, **VoIP.ms** offers unbeatable pricing and a feature-rich DIY portal, though with the caveat of more self-management. For organizations that demand **enterprise-grade reliability and global reach**, **Telnyx** and **Twilio** stand out: Telnyx often wins on price and personalized support, whereas Twilio offers an unparalleled global platform and ecosystem of services (at a higher price). **Flowroute** remains a strong contender for North America-centric deployments needing high call quality, no-nonsense pricing, and proven interoperability.

From a technical decision-maker's perspective, it is wise to evaluate a shortlist of 2–3 providers with a trial. For instance, you might start a trial with ClearlyIP (to test integration ease and support responsiveness), Telnyx (to evaluate global call quality and features like encryption), and VoIP.ms (to gauge cost savings and portal functionality). During the trial, measure key metrics: call setup time, audio quality (try different codecs), failover behavior, and support response to any queries. Also consider future needs: if you plan to expand internationally, lean toward a provider with broad coverage (Telnyx, Twilio). If you need tight SIP integration with custom apps or are already using cloud communications APIs, Twilio or SignalWire might fit well. If you operate in a regulated industry, ensure the provider can meet compliance (most do, but Twilio and Telnyx have explicit certifications and offerings for that).

Security should not be an afterthought – ensure your chosen provider supports TLS/SRTP if you require encrypted calling. According to OnSIP's blog, **"you should make sure that your VoIP provider has [TLS and SRTP]"** onsip.com, and as shown above, the leading providers do support these.

Finally, consider using more than one provider for redundancy. Asterisk allows configuring multiple trunk peers, so some integrators use, for example, ClearlyIP as primary and Flowroute or Twilio as backup (or vice versa). This can protect against provider-specific outages and also let you take advantage of strengths (maybe one has cheaper international rates, another has better domestic coverage).

In summary, the VoIP providers detailed in this report represent the top options for Asterisk PBX systems in 2025. Each excels in certain areas:

- **ClearlyIP:** Best for FreePBX integration and balanced features with strong support community.freepbx.org.
- **SIPStation (Sangoma):** Best for all-in-one Asterisk solution with flat fee simplicity.
- **Flowroute:** Best for pay-per-use flexibility with high voice quality flowroute.com.
- **VoIP.ms:** Best for budget-conscious users who want control and lots of features.
- **Telnyx:** Best for global deployments needing advanced features, low latency, and top-notch support telnyx.comsip.telnyx.com.

- **Twilio:** Best for large-scale and multi-country integration, especially if leveraging other Twilio services research.comresearch.com.

By carefully assessing the criteria in this report – from interoperability and codecs to pricing and reliability – businesses and integrators can confidently select a SIP trunk provider that aligns with their Asterisk system requirements and business goals. Remember to periodically re-evaluate providers as the VoIP landscape evolves (for example, new codec support or pricing changes) to ensure you continue to have the best fit for your Asterisk PBX.

Tags: asterisk, pbx, voip, sip trunking, telephony, codecs, security, reliability, clearlyip, freepbx

About ClearlyIP

ClearlyIP Inc. — Company Profile (June 2025)

1. Who they are

ClearlyIP is a privately-held unified-communications (UC) vendor headquartered in Appleton, Wisconsin, with additional offices in Canada and a globally distributed workforce. Founded in 2019 by veteran FreePBX/Asterisk contributors, the firm follows a "build-and-buy" growth strategy, combining in-house R&D with targeted acquisitions (e.g., the 2023 purchase of Voneto's EPlatform UCaaS). Its mission is to "design and develop the world's most respected VoIP brand" by delivering secure, modern, cloud-first communications that reduce cost and boost collaboration, while its vision focuses on unlocking the full potential of open-source VoIP for organisations of every size. The leadership team collectively brings more than 300 years of telecom experience.

2. Product portfolio

- **Cloud Solutions** – Including *Clearly Cloud* (flagship UCaaS), **SIP Trunking**, **SendFax.to** cloud fax, **ClusterPBX OEM**, **Business Connect** managed cloud PBX, and **EPlatform** multitenant UCaaS. These provide fully hosted voice, video, chat and collaboration with 100+ features, per-seat licensing, geo-redundant PoPs, built-in call-recording and mobile/desktop apps.
- **On-Site Phone Systems** – Including CIP PBX appliances (FreePBX pre-installed), ClusterPBX Enterprise, and Business Connect (on-prem variant). These offer local survivability for compliance-sensitive sites; appliances start at 25 extensions and scale into HA clusters.
- **IP Phones & Softphones** – Including CIP SIP Desk-phone Series (CIP-25x/27x/28x), fully white-label branding kit, and *Clearly Anywhere* softphone (iOS, Android, desktop). Features zero-touch provisioning via Cloud Device Manager or FreePBX "Clearly Devices" module; Opus, HD-voice, BLF-rich colour LCDs.
- **VoIP Gateways** – Including Analog FXS/FXO models, VoIP Fail-Over Gateway, POTS Replacement (for copper sun-set), and 2-port T1/E1 digital gateway. These bridge legacy endpoints or PSTN circuits to SIP; fail-over models keep 911 active during WAN outages.
- **Emergency Alert Systems** – Including **CodeX** room-status dashboard, **Panic Button**, and **Silent Intercom**. This K-12-focused mass-notification suite integrates with CIP PBX or third-party FreePBX for Alyssa's-Law compliance.
- **Hospitality** – Including **ComXchange** PBX plus PMS integrations, hardware & software assurance plans. Replaces aging Mitel/NEC hotel PBXs; supports guest-room phones, 911 localisation, check-in/out APIs.
- **Device & System Management** – Including **Cloud Device Manager** and **Update Control (Mirror)**. Provides multi-vendor auto-provisioning, firmware management, and secure FreePBX mirror updates.
- **XCast Suite** – Including Hosted PBX, SIP trunking, carrier/call-centre solutions, SOHO plans, and XCL mobile app. Delivers value-oriented, high-volume VoIP from ClearlyIP's carrier network.

3. Services

- **Telecom Consulting & Custom Development** – FreePBX/Asterisk architecture reviews, mergers & acquisitions diligence, bespoke application builds and Tier-3 support.
- **Regulatory Compliance** – E911 planning plus **Kari's Law**, **Ray Baum's Act** and **Alyssa's Law** solutions; automated dispatchable location tagging.
- **STIR/SHAKEN Certificate Management** – Signing services for Originating Service Providers, helping customers combat robocalling and maintain full attestation.
- **Attestation Lookup Tool** – Free web utility to identify a telephone number's service-provider code and SHAKEN attestation rating.
- **FreePBX® Training** – Three-day administrator boot camps (remote or on-site) covering installation, security hardening and troubleshooting.
- **Partner & OEM Programs** – Wholesale SIP trunk bundles, white-label device programs, and ClusterPBX OEM licensing.

4. Executive management (June 2025)

- **CEO & Co-Founder: Tony Lewis** – Former CEO of Schmooze Com (FreePBX sponsor); drives vision, acquisitions and channel network.
- **CFO & Co-Founder: Luke Duquaine** – Ex-Sangoma software engineer; oversees finance, international operations and supply-chain.
- **CTO & Co-Founder: Bryan Walters** – Long-time Asterisk contributor; leads product security and cloud architecture.
- **Chief Revenue Officer: Preston McNair** – 25+ years in channel development at Sangoma & Hargray; owns sales, marketing and partner success.
- **Chief Hospitality Strategist: Doug Schwartz** – Former 360 Networks CEO; guides hotel vertical strategy and PMS integrations.
- **Chief Business Development Officer: Bob Webb** – 30+ years telco experience (Nsight/Cellcom); cultivates ILEC/CLEC alliances for Clearly Cloud.
- **Chief Product Officer: Corey McFadden** – Founder of Voneto; architect of EPlatform UCaaS, now shapes ClearlyIP product roadmap.
- **VP Support Services: Lorne Gaetz** (appointed Jul 2024) – Former Sangoma FreePBX lead; builds 24×7 global support organisation.
- **VP Channel Sales: Tracy Liu** (appointed Jun 2024) – Channel-program veteran; expands MSP/VAR ecosystem worldwide.

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5. Differentiators

- **Open-Source DNA:** Deep roots in the FreePBX/Asterisk community allow rapid feature releases and robust interoperability.
- **White-Label Flexibility:** Brandable phones and ClusterPBX OEM let carriers and MSPs present a fully bespoke UCaaS stack.
- **End-to-End Stack:** From hardware endpoints to cloud, gateways and compliance services, ClearlyIP owns every layer, simplifying procurement and support.
- **Education & Safety Focus:** Panic Button, CodeX and e911 tool-sets position the firm strongly in K-12 and public-sector markets.

In summary

ClearlyIP delivers a comprehensive, modular UC ecosystem—cloud, on-prem and hybrid—backed by a management team with decades of open-source telephony pedigree. Its blend of carrier-grade infrastructure, white-label flexibility and vertical-specific solutions (hospitality, education, emergency-compliance) makes it a compelling option for ITSPs, MSPs and multi-site enterprises seeking modern, secure and cost-effective communications.
