LiveSwitch

White Paper

Making Sense of SIP, Telephony, VoIP & PTSN

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Telephony

Making sense of WebRTC, SIP, VoIP, PBXs and the PSTN

Introduction

Telephony integration is an important part of a fully-connected architecture for Internet-based real-time communication projects. While web browsers and native apps tend to be the primary focus of a modern software project, the ubiquity and simplicity of a simple mobile or wired telephone makes it a powerful option for connecting people.

Public Switched Telephone Network (PSTN)

When talking about telephony, we have to start with the public switched telephone network (PSTN), sometimes referred to as the plain old telephone service, or POTS. The PSTN is essentially the Internet for telephones. It's the network of systems that connect all the telephones across the world, both wired and wireless, making it possible to connect with someone by simply dialling a number.

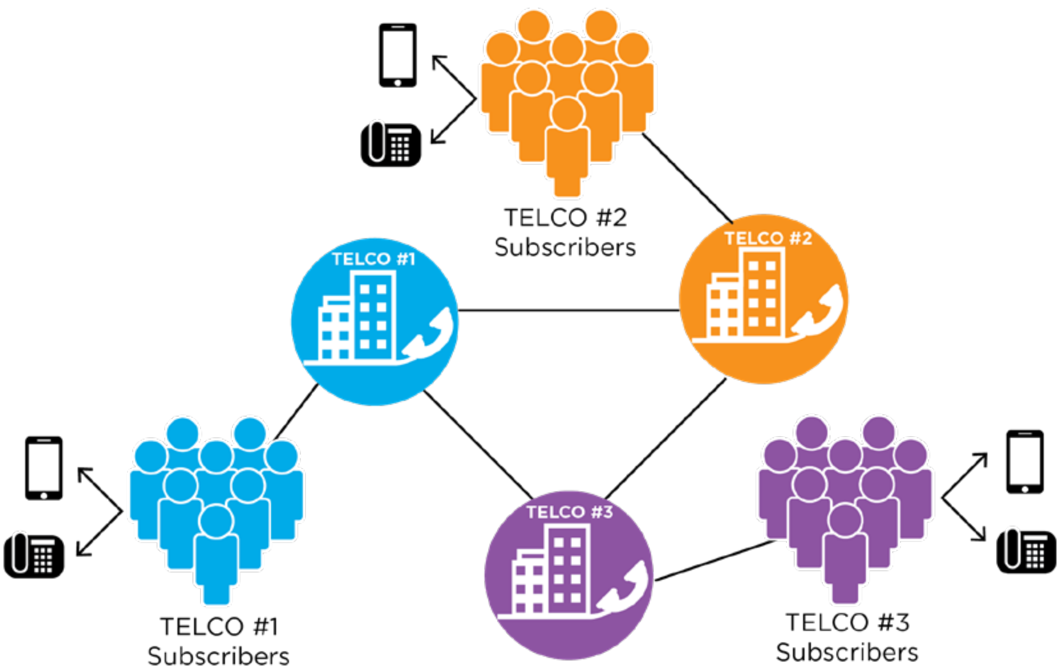

Fig 1: Public Switched Telephone Network (PSTN)

Private Branch Exchanges (PBXs)

Separate from the PSTN are private telephone networks, known as private branch exchanges (PBX). A PBX is typically used in business settings where you would want to be able to dial an extension to reach someone else within the private network. Most PBXs allow outbound calling to the PSTN, known as "termination", which allow devices on the private network to dial public numbers that get routed out to the public network. In many cases, the PBX will also support inbound calling from the PSTN, known as "origination", which allow devices on the private network to receive calls from the public network, either by a direct number or through the use of an intermediary answering system that lets the caller provide an extension.

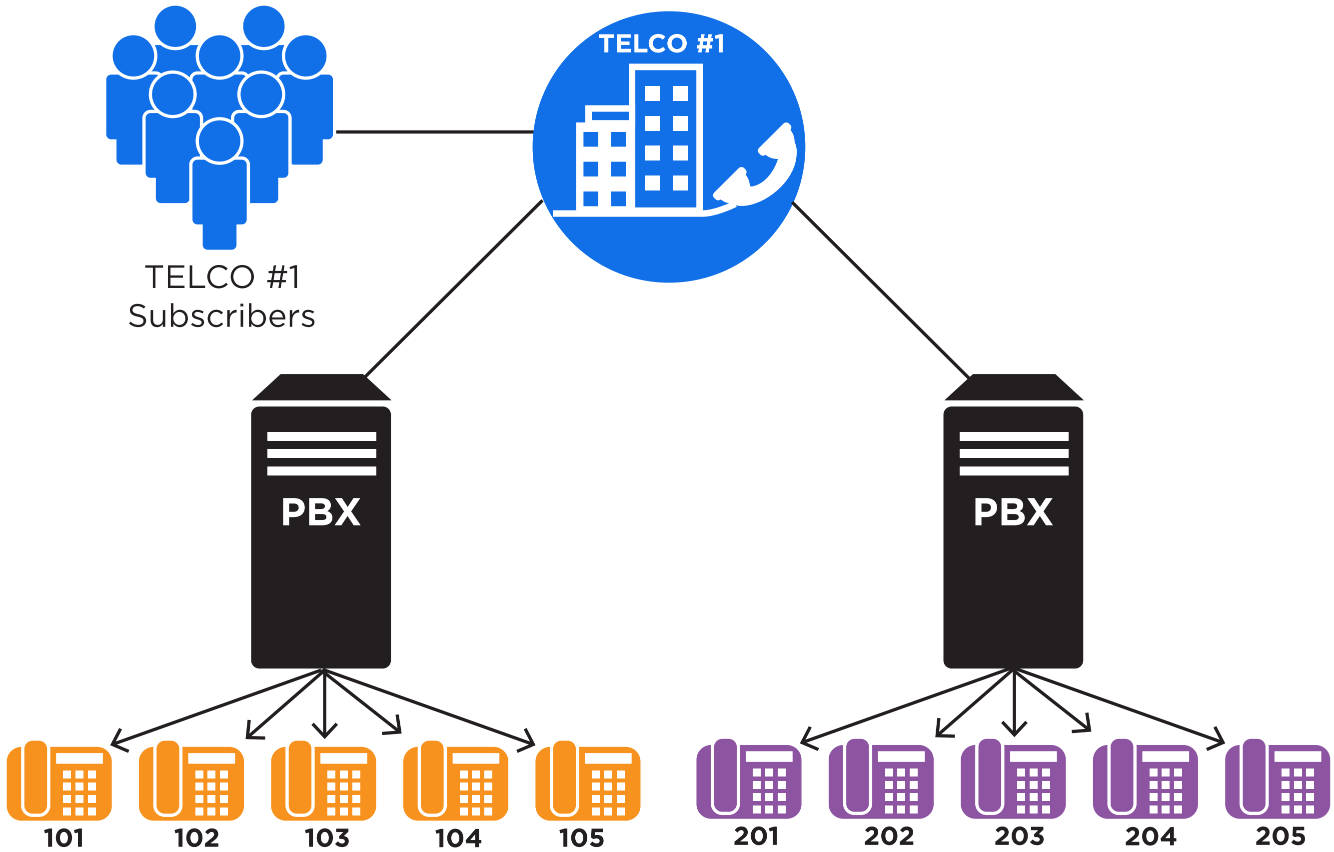

Fig 2: Private Branch Exchanges (PBX) connected to PSTN

Direct Inward Dialing

Direct numbers for dialling into a PBX are known as direct inward dialing (DID) or direct dial-in (DDI) numbers and must be purchased from the telephone service provider. Using an intermediary answering system, known as an interactive voice response (IVR) system is generally less expensive as it allows a single DID/DDI number to route to many devices on the private network, but requires the caller to interact with the IVR system so it knows where to route the call.

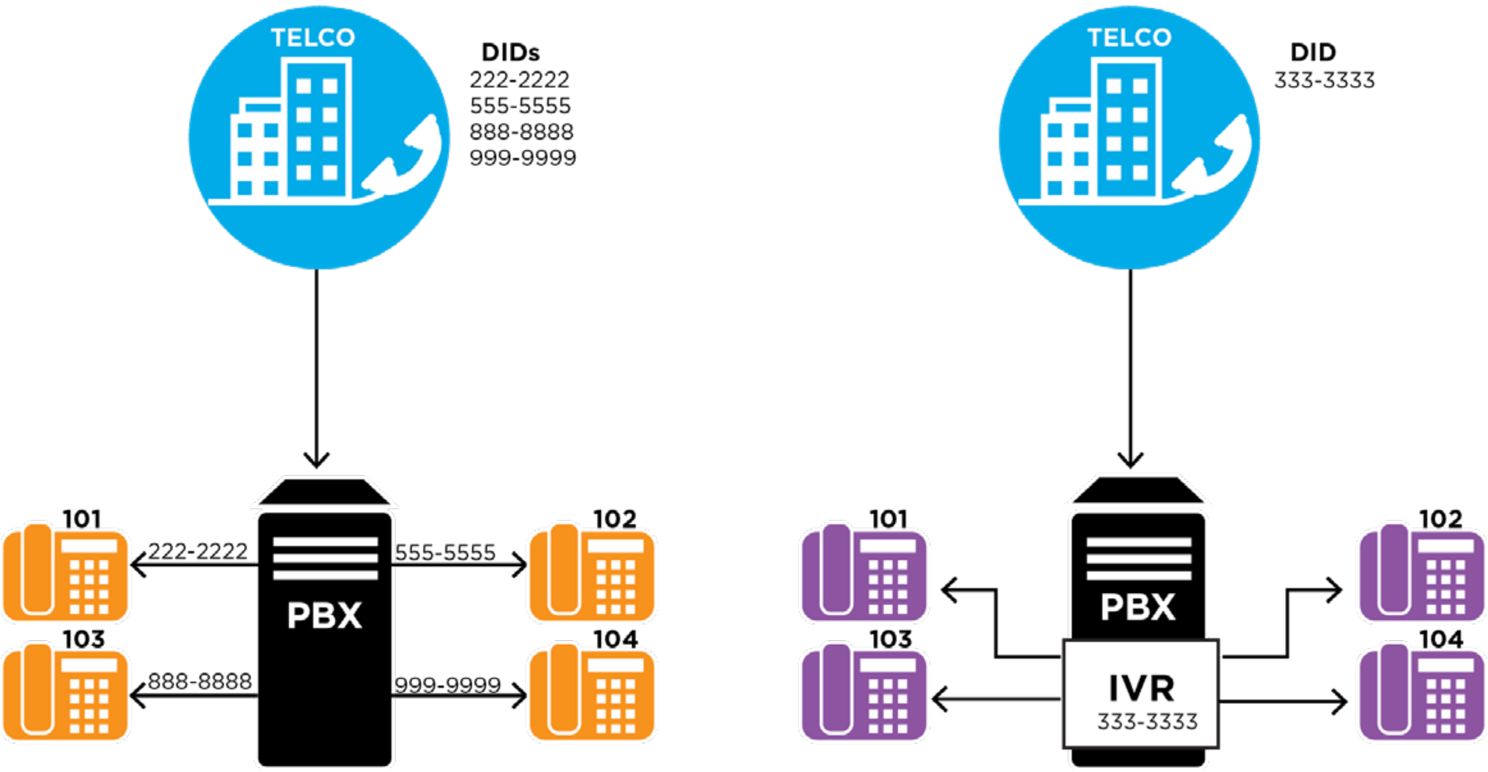

Fig 3: Direct Inward Dialing (DID) usage with and without an IVR

Circuit Switching

Historically, both the PSTN and PBXs relied on circuit switching. In a circuit-switched network, when two devices talk to each other, they do so over a dedicated, unbroken wire physically connecting them. Calls placed over circuit-switched networks can make excellent guarantees on quality of service, since each call has dedicated physical hardware supporting it without competition from any other devices. They are, however, expensive to build and maintain. They are also generally inefficient since any unused room on the wire cannot be shared, although time-division multiplexing (TDM) makes this somewhat better.

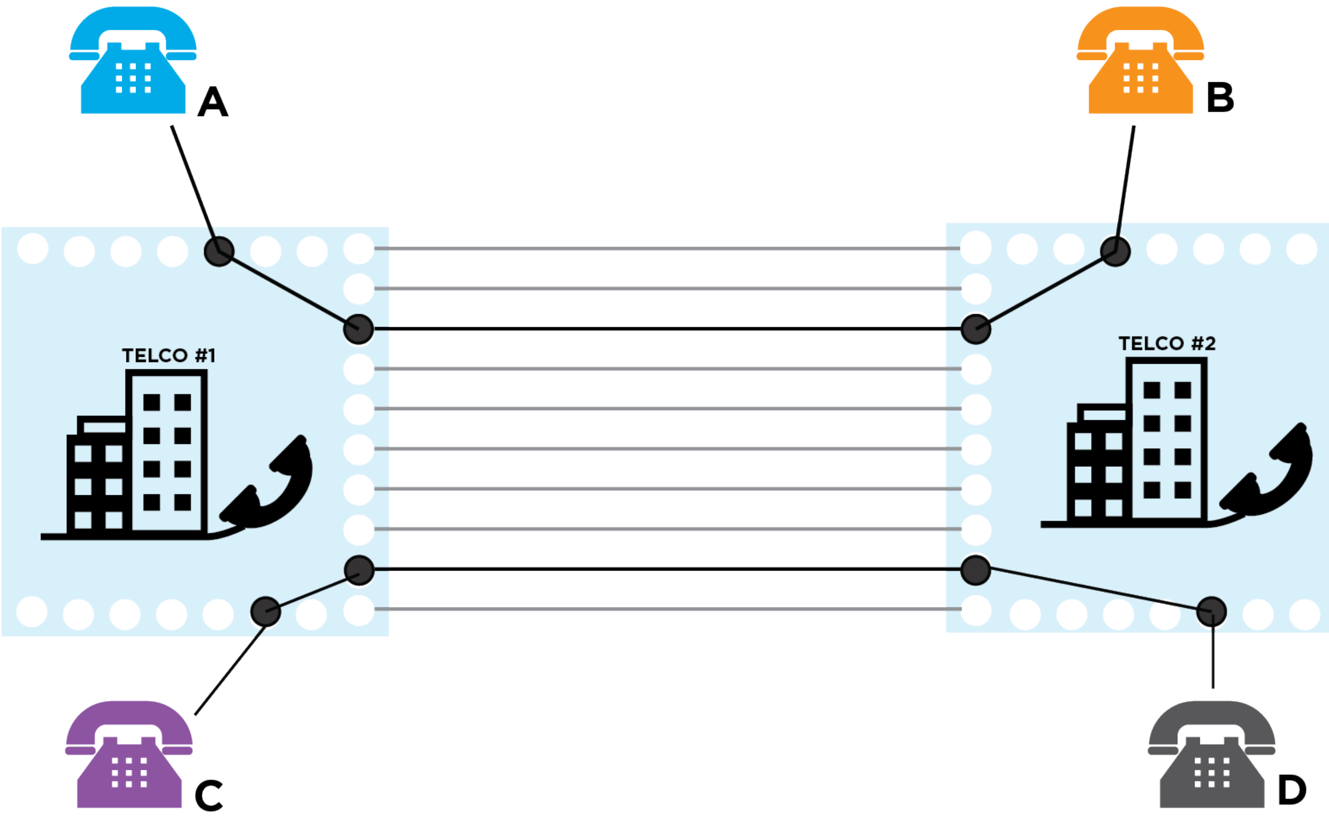

Fig 4: Historical circuit switching telephony

Packet Switching

The Internet, on the other hand, relies on packet switching. Packet switching uses a shared network approach, where the physical wires are shared among a large number of devices. Data messages sent on a packet-switched network use headers to indicate source and destination addresses so the traffic can be routed to the correct location. While packet-switched networks cannot always make the same quality of service guarantees as their circuit-switched counterparts, they operate much more efficiently and with a significantly lower price tag.

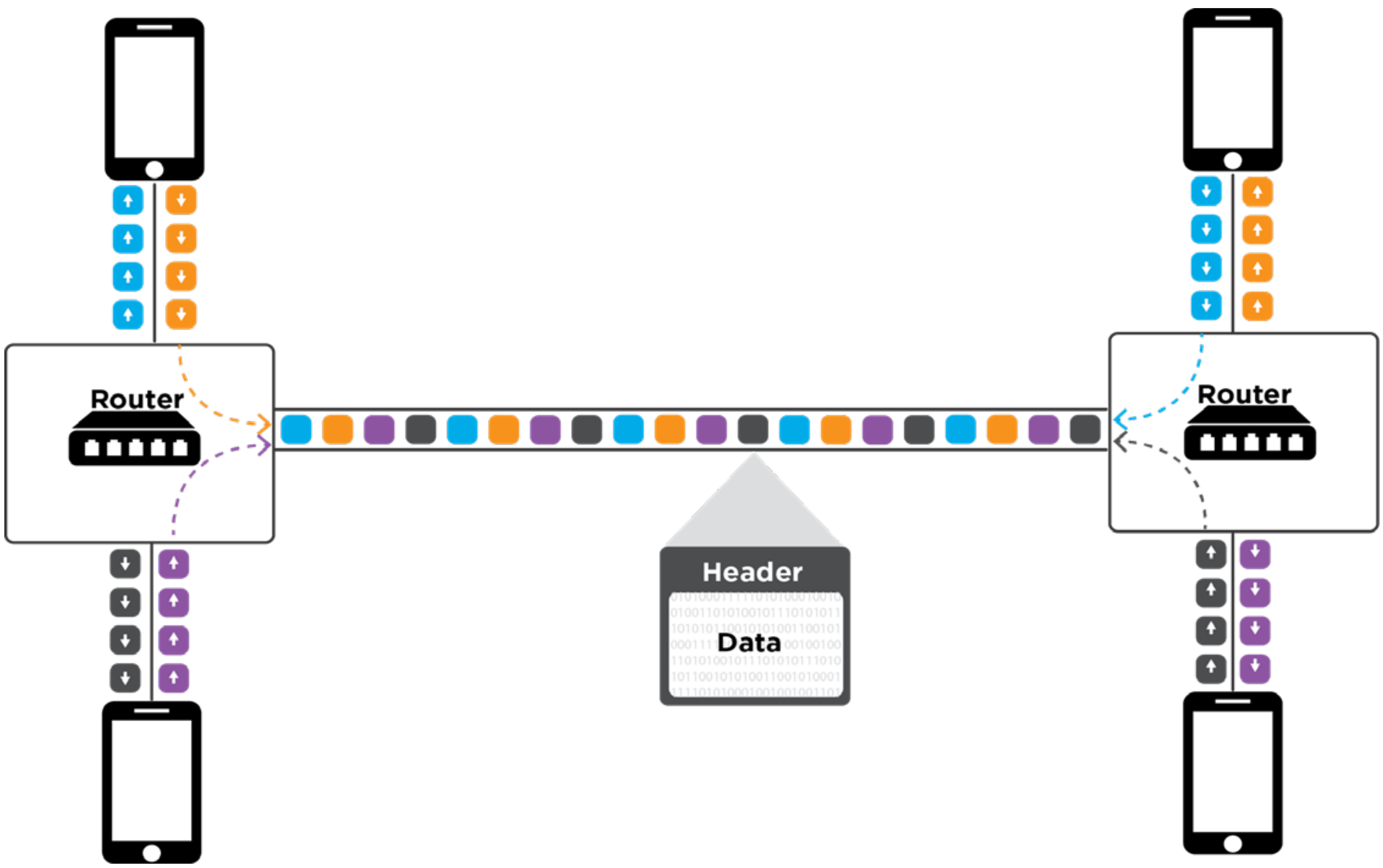

Fig 5: Packet Switching Telephony

Voice over IP (VoIP)

With the emergence of the Internet came the concept of Voice over IP (VoIP), which sends and receives audio and sometimes video over packet-switched networks (the Internet typically) instead of circuit-switched networks. This made connecting a PBX to the PSTN much less expensive. Instead of needing a dedicated T1/E1 line (also called a trunk) and special hardware, it was possible to use existing Internet lines and commodity hardware with an IP PBX software package. SIP trunking providers emerged as the bridge between these new IP-based systems and the legacy circuit-switched networks (more on SIP below). Today, all you need to create a PBX is a server to run your software (e.g. FreeSWITCH, Asterisk, or 3CX), and an account with a SIP trunking provider (e.g. Flowroute, Voxbone, or Thinktel).

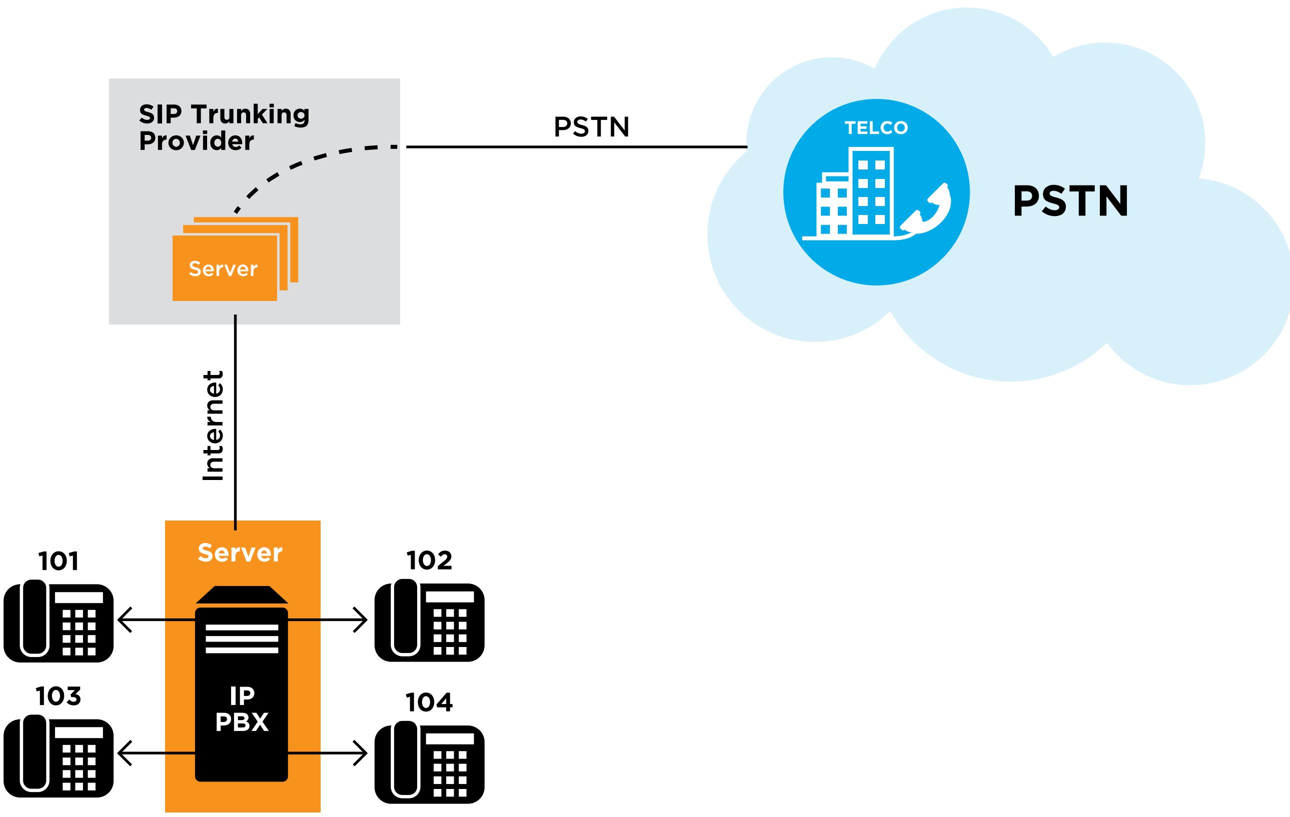

Fig 6: SIP Trunk Connection to PSTN

Session Initiation Protocol (SIP)

The primary signalling protocol used by VoIP systems when establishing calls is the session initiation protocol, or SIP for short, and is the protocol used by LiveSwitch to connect telephone devices to its media server. Older call signalling technologies like H.323 still exist, but have largely been supplanted by SIP or had their deployments augmented with gateways that bridge between the legacy protocol and SIP.

SIP trunking providers use the SIP protocol to talk over the Internet to your SIP-based VoIP devices (PBX, softphone, etc.). The SIP Connector included with LiveSwitch uses the SIP protocol together with your SIP trunk account credentials to place outbound calls, route audio and video to and from the LiveSwitch media server, and receive inbound calls on DID/DDI numbers purchased from your provider.

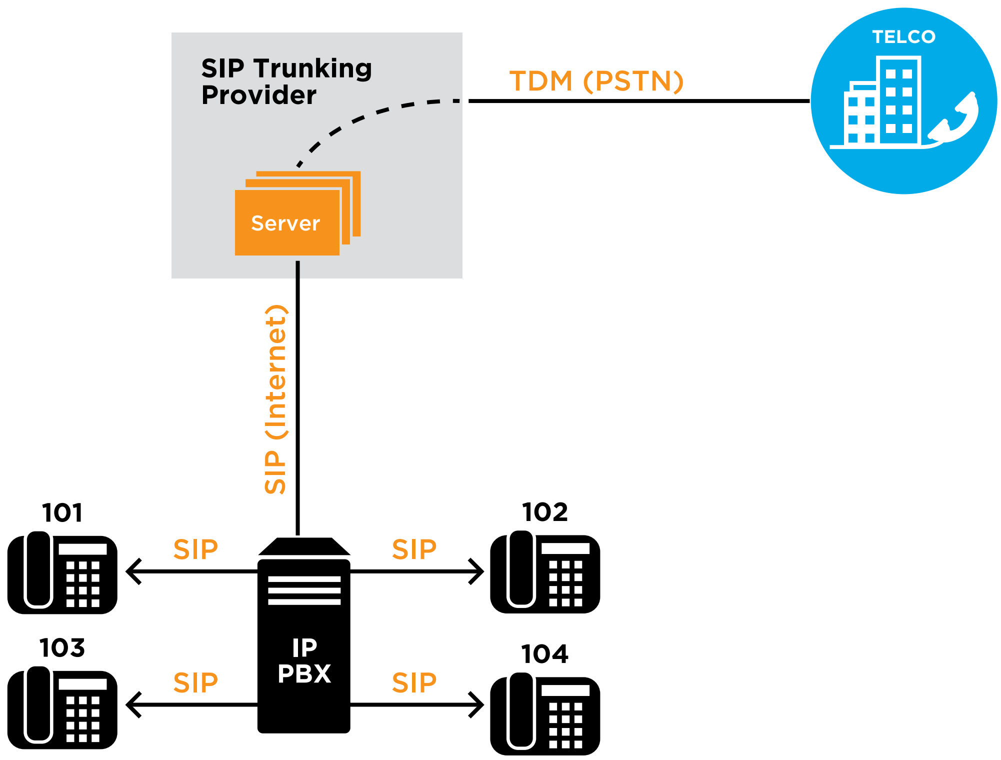

Fig 7: SIP Signaling

WebRTC

WebRTC uses many of the same underlying protocols that VoIP uses (i.e. RTP, RTCP, SRTP, SDP), but deliberately excludes SIP, and all other the signalling protocols for that matter, from the specification. Instead, the choice of how to signal for call establishment is left to the developer. This opens up an infinite range of possible use cases and avoids being tied to the murky complexity of the SIP protocol and its vast array of extensions, but makes interoperability between WebRTC-based and SIP-based devices much more challenging.

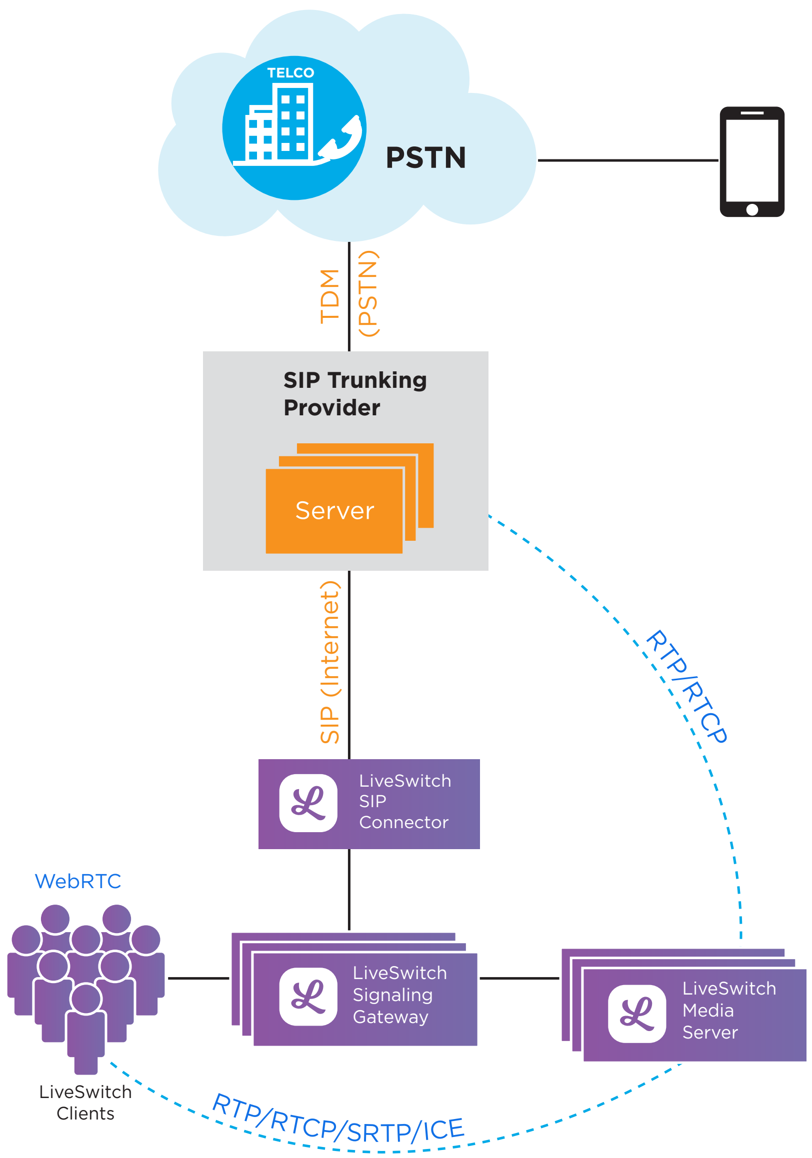

Fig 8: Connecting a WebRTC Client to the PSTN

The SIP Connector included with LiveSwitch takes care of this challenge. Inbound and outbound phone calls are handled with the SIP protocol and are bridged seamlessly with the application logic and media flow of your front-end web and native applications. Bring your own SIP trunk and the rest is handled.

WebRTC also forces a high level of security and network efficiency by taking some pieces of the various protocols and making them mandatory or "mandatory-by-default" (overridable in code or by using command-line switches). This is good for the privacy and overall experience of the end user, but again, makes telephony integration more challenging as individual SIP providers may or may not support these mandatory components, or may have requirements of their own that are incompatible with vanilla WebRTC.

Again, the SIP Connector included with LiveSwitch handles this with its very design. It is custom-tailored to meet the requirements of VoIP services, ensuring that calls always succeed. The media flow to and from each telephone is uniquely handled independently of all other connections using the LiveSwitch mixer (MCU), ensuring that security and speed is maximized wherever possible without compromising on interoperability. Any special SIP-related firewall rules only need be set up once on the server side.

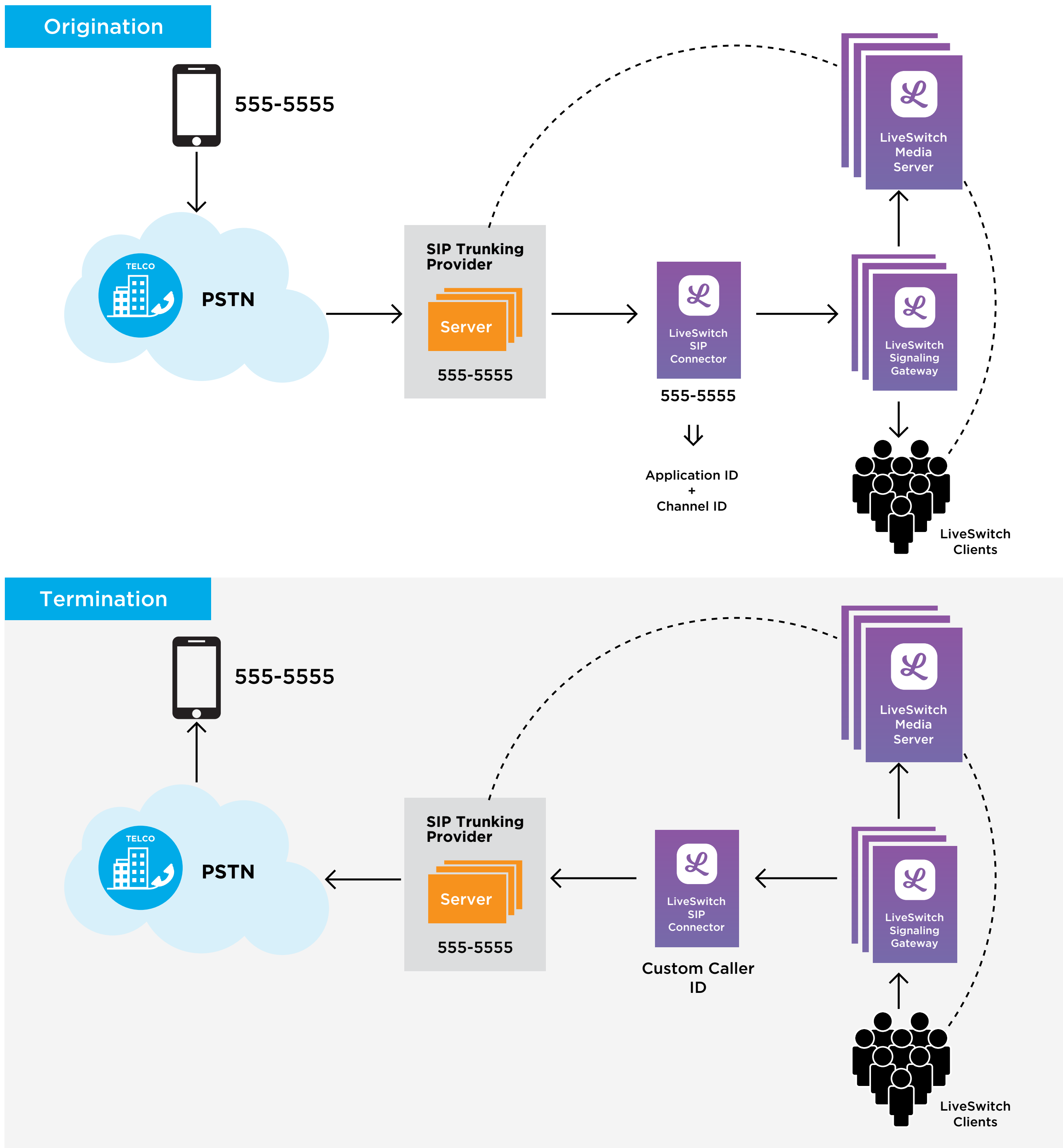

Fig 9: Connecting a WebRTC Client to the PSTN, Origination & Termination

Wrap-UP

LiveSwitch makes integrating with telephony systems simple and scalable. The included SIP Connector bridges the gap between WebRTC and the PSTN, leveraging the power of both the LiveSwitch media server and third-party SIP trunk services to interoperate with telephony services. Whether you want your application to allow simple point-to-point calling, dialling into conferences, or invitations to ongoing conferences, LiveSwitch can make it happen quickly and reliably.

If you are interested in learning more, don't hesitate to contact us. Our product suite is custom-designed from the ground up to be flexible enough to work in every scenario for every customer, regardless of how unique or constrained your needs are, and yet powerful and scalable enough to serve massive customer bases. We look forward to hearing from you!