VanillaIP SIP Trunking

At its simplest, SIP Trunking provides ISDN replacement into existing PABX's to provide cheaper access charges for the customer.

SIP Trunking is about more than cost savings on access. We see it is a critical element in unifying all elements of the customers telecoms infrastructure. This provides a homogenous communications strategy across all the customers' platforms and sites and retains existing investment.

VanillaIP SIP Trunking extends cost reduction by overlaying next generation Intelligent Network services. This includes sophisticated queuing, inbound call statistics, centralised reception, remote working, distributed call centres and desk-to-desk dialling.

Each SIP Trunk supports one concurrent external call, equivalent to one ISDN channel. For most customers this represents a saving of up to 75% compared to ISDN. All elements of the customer's numbers and call routing policy are unchanged. In this way you can VoIP enable any PBX and provide your customers a community of users rather than individual islands.

A major benefit for resellers is the ability to bring together different sites using any mixture of SIP Trunking and Hosted PBX Extensions. All users become available via their 4 digit extension number from any other extension, without the need for QSig, DPNSS or V-LANS's. This combines the customer numbering plans together like a Voice-VPN.

Inbound Calling Options

Number Porting – Retain existing phone numbers. We can port numbers from every carrier

Extension to Extension – SIP Trunk extensions can call a hosted PABX extension in the same or another office, just by dialling the extension number

Multi Mixed Extensions – Users that have an office extension from the existing PABX, but want a home phone can use Shared Call Appearance. This provides multiple phones 'twinned' on the same extension/DDI number

CLI Presentation – Any extension user is able to present either a group CLI or Individual DDI

Call Queuing – Queue callers in the VanillaIP core further reducing the number of channels required locally

Call Recording – Can be activated for PABX extensions on inbound and outbound calls

Non Geographic Numbers – Provide 07,08,09 numbers as 'logical' numbers without any divert to a geographic DDI

International Numbers – Supply your customers with international numbers from over 35 countries. Connect direct with no diversion charges

vanillaIP
joined up thinking

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SIP Trunking offers a number of advantages over traditional ISDN:

Installation Line Cost Reduction – The installation cost for SIP Trunking is up to 80% cheaper than ISDN installation costs.

Increasing Capacity – Where an existing PBX is at capacity, SIP trunking allows for deployment of hosted extensions alongside the existing system. All users can have a common numbering plan and desk-to-desk dialling.

Linking Remote Sites or Home Workers – SIP Trunking allows Remote Workers to be integrated within the main office with on-net calling and group service membership, without the complexity of tunnelling service (Q-sig, DPNSS, VPN's) out from the PBX.

Call Queuing and Statistics – Because inbound calls queue in the network core, there is no need to stack ISDN locally just to provide queuing and comfort messages to callers. For inbound call centres, which typically have a higher number of ISDN channels than Agents, this represents a massive cost saving. We can also provide statistics on the volume and pattern of incoming calls with abandoned ratios, average wait/talk times etc.

UK Geographic Numbers – SIP Trunking allows your London office to have a Manchester number delivered directly. This allows you to publish local numbers around the UK without having to pay diversion charges back to the main office.

Connectivity Options

Any PBX that can accept ISDN/Q.931 can accept VanillaIP SIP Trunks, either through native SIP support on newer platforms or through an external gateway on older systems.

Native SIP Interface Support – New PBX systems with SIP support can terminate the ADSL directly onto the PABX chassis.

The gateway supports protocol – Older, non-SIP, PBX's require a gateway to connect the ADSL with SIP Trunks. The gateway supports protocol conversion from SIP to Q.931. In this way the existing ISDN/Q.931 trunk card on the PBX connects to the gateway, so the customer does not have additional PBX hardware costs. The gateway dial plan will also allow extension dialling from PBX extensions to other VanillaIP hosted VoIP extensions.

Standard vs Enhanced SIP Trunks

Standard SIP Trunks – Standard SIP trunks carry outbound call traffic.

Enhanced SIP Trunks – Enhanced SIP Trunks carry both inbound and outbound call traffic and can also use optional VanillaIP Intelligent Network Services, such as Call Centre, cloud queuing and inbound call stats. There is no limit to the number of PABX extensions and DDI's that can be supported behind Enhanced SIP Trunks.

SIP Trunk Group – A SIP Trunk Group is also required whenever SIP trunks are deployed.

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