VANILLAIP SIP TRUNKING

VANILLAIP SIP TRUNKING PROVIDES ISDN REPLACEMENT INTO EXISTING PBX SYSTEMS FOR ACCESS CHARGE COST REDUCTION AND INTELLIGENT NETWORK SOLUTIONS FROM THE VANILLAIP CLOUD

SIP Trunking provides more than cost savings on access. We see it is a critical element in unifying all elements of your telecoms infrastructure. This provides a homogenous communications strategy across all platforms and sites and retains existing investment. All existing numbers and call routing policies are unchanged

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

A major benefit is the ability to bring together different sites using any mixture of SIP Trunking and Hosted PBX Extensions. All users become available via their extension number from any other extension for maximum usability. In this way you can VanillaIP Cloud enable any PBX and provide customers with a community of users rather than individual islands.

FEATURE OVERVIEW

MULTI-SITE BURSTING

A single customer group can be assigned a “master” number of concurrent SIP Trunks which can then be allocated and oversubscribed by their sites. For example, the group can have 40 SIP Trunks and each of their three sites can be assigned 20 SIP Trunks each. In this example, each site can dynamically use any number up to the allocated 20x20, provided the cumulative total does not exceed 40. Service Providers can charge for this service either based on the 40 at the Group level or the 60 total assigned at the Sites level. This bursting provides total flexibility and maximises ROI for multi-site customers

MOBILITY BOLT-ON

The hosted PABX user service “Mobility Bolt-On” can now be applied to legacy PBX users, empowering them with current UC features including Sim Ring, VanillaIP Anywhere and Remote Office. This provides a powerful fixed-mobile upsell for PBX extensions on the back of VanillaIP SIP Trunking

1 DAY RENTALS

To cater for short-term demand spikes, customers can now increase or decrease SIP Trunk channel capacity on-demand and only pay for service as they consume it. SIP Trunks are available on 1 day rentals, with no 30 day notice period making the service ideal for seasonal customers or those whose workload is highly project based.

DISASTER RECOVERY

Our DR capability applies equally across Hosted PBX and SIP Trunking allowing a pre-set default call routing behaviour to be invoked when disaster strikes. Individual DDI numbers, hunt groups and queues can all be diverted at a Group or Site level with a single click.

CALL LOGS AND CREDIT LOCKING

Credit locks are available at the group level [No SIP User service on the PABX extension] and also to the PABX extension level with either SIP User Basic or Enhanced. Credit Locking will automatically bar any group or extension that has exceeded the threshold in Uboss. Email alerts will notify of any user spend that is nearing the threshold.

INTERNATIONAL NUMBERS AND INTERNATIONAL DEPLOYMENTS

International Numbers can now be assigned through Uboss in the same way as geo and non-geo numbers as a "logical" number on the platform, without having to re-point to an underlying DDI number. Partners can also deploy SIP Trunks abroad.

SINGLE DIAL PLAN

Because all the smarts are in the cloud you get extension dialling across sites by default and the ability to intelligently link any mix of traditional PBX, Hosted PBX sites, home and mobile workers across your telecoms estate.

CLOUD QUEUING

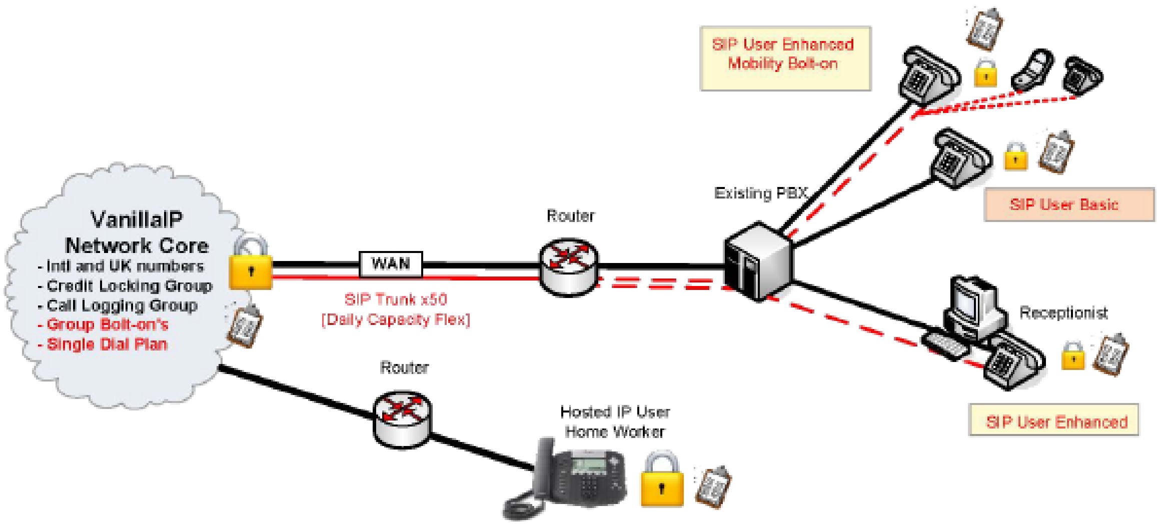
Because inbound calls queue in the network core, there is no need to stack ISDN locally just to provide queuing and comfort messages to callers. For inbound call centres, which typically have a higher number of ISDN channels than Agents, this represents a massive cost saving.

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 simply and easily.

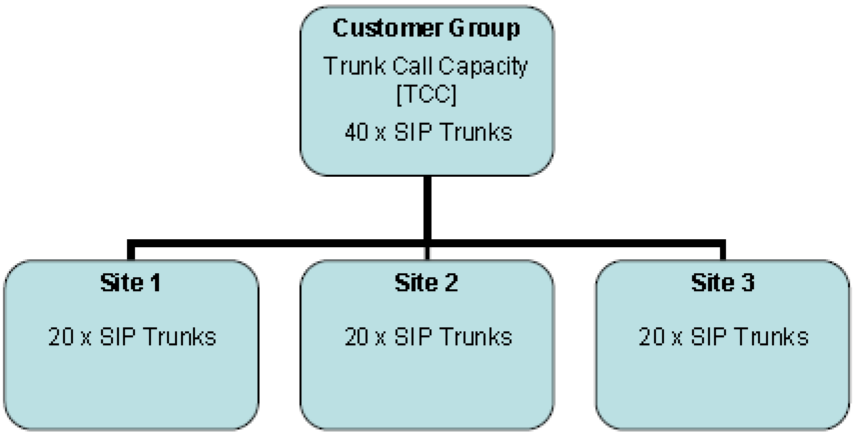


INBOUND CALLING OPTIONS

- NUMBER PORTING
  Retain existing phone numbers. We can port numbers from every carrier
- EXTENSION TO EXTENSION
  SIP Trunk extensions can call a hosted PBX extension in the same or another office, just by dialling the extension number
- MIXED HOSTED AND PBX EXTENSIONS
  Users that have an office extension from the existing PBX, 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 Technology core further reducing the number of channels required locally
- CALL RECORDING
  Can be activated for PBX extensions on inbound and outbound calls
- NON GEOGRAPHIC NUMBERS
  Provide 03, 07, 08, 09 numbers as 'logical' numbers without any divert to a geographic DDI
- INTERNATIONAL NUMBERS
  Choose numbers from over 35 countries, most with specific city codes

LICENSING OVERVIEW

- SIP Trunks are allocated as a Trunk Call Capacity [TCC] at the customer Group level. This is the total numbers of SIP Trunks/channels that can be used by the customer.
- Individual sites can now be allocated any number of SIP trunks up to the total TCC at the Group level. For example, a Group could have 40 x SIP Trunks at the TCC level and three sites with 20 x SIP Trunks each. This allows customers to "burst" local capacity up to the total TCC level
- Service Provider partners can choose to bill trunks at the TCC or individual Site level. This latter option provides an option to maximise and upsell revenue. VanillaIP billing is based on the number of SIP Trunks allocated at the TCC



BUSINESS (GROUP) AND USER PACKS AND SERVICES

SERVICE DESCRIPTION	USER TYPE		
	NO SIP USER	SIP USER BASIC	SIP USER ENHANCED
Trunking Extn to Hosted Extn dialling		•	•
Credit Lock - Group	•	•	•
Credit Lock – User		•	•
Call Barring – Group	•	•	•
Call Barring – User		•	•
Call Logging - Group	•	•	•
Call Logging – User		•	•
Direct Dial Number		Optional Add-On	Optional Add-On
Unified VoiceMail		Optional Add-On	Optional Add-On
Mobility Bolt-On		Optional Add-On	Optional Add-On
Fax Messaging		Optional Add-On	Optional Add-On
Inbound Call Recording		Optional Add-On	Optional Add-On
Auto Attendant		Optional Add-On	Optional Add-On
CLI Outbound – Group	•	Optional	Optional
CLI Outbound – User		Optional	Optional
CLI Outbound – Restrict – User		Optional	Optional
Alternate Number			•
Call Forward Diverts			•
Call Reject – Anonymous			•
Call Selective Acceptance			•
Call Selective Rejection			•
Disaster Recovery - User			•

SERVICES BY SIP USER PACKAGE

BUSINESS LEVEL	**Trunk Call Capacity (TCC)** • Charge Per Channel (concurrent call) • No minimum quantity • Can be varied on a daily basis • No authentication required if user credit locking being use •
SITE SERVICE	**Trunk Group** • Multiply Trunk Groups across different sites allowed • No charge for Trunk Groups • Maximum number of con-current calls determined by TCC • Each Trunk Group must have at least 1 x SIP User – Basic, which is the pilot user
USER SERVICES (PABX EXTENSION)	**No SIP User** • Inbound call to Reception only • No User allocated within Uboss to underlying PABX Extension • Outbound Dialling Only • Credit Lock – Business Level – calls rated within 30 minutes of call completion • Call Bar Outbound - Group • Call Logs - Group
	SIP User - Basic • Ext. dial between SIP and Hosted Ext. within business group • Credit Lock – User Level • Call Barring – User Level • Call Logs – down to extension number • DDI – Direct Dial Number • CLI – Outbound presentation – present any number owned by business • CLI – Outbound Restriction – withhold CLI on call by call basis or for all calls • CLI – Internal Presentation – show own CLI on internal calls
	BoltOn-SIP User – Enhanced [can be added to SIP User Basic] *As SIP User - Basic above but also includes* • Alternate Number – Up to 10 inbound number can be point to the user, Geo, Non Geo, International • Call Divert - Always, Busy, No Answer, Not Reachable • Call Reject [Anonymous, Selective