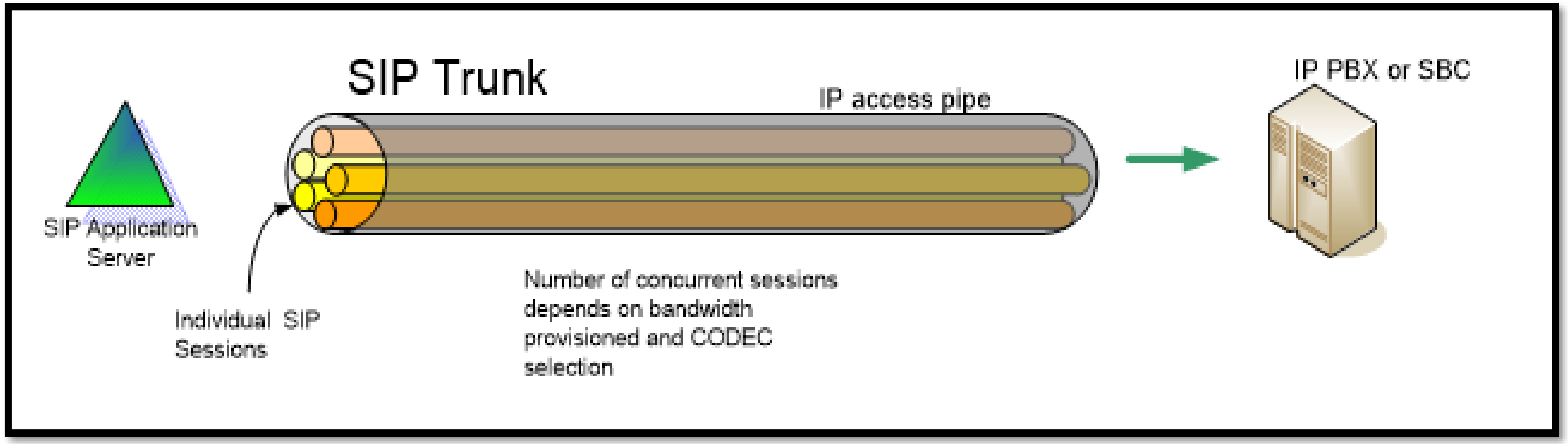
# SIP TRUNKING

# What is SIP Trunking?




- A SIP “Trunk” connects an IP PBX to the Telkom VoIP network to enable voice calls
- **Made up of 3 components;**

1. Access – This is the data connectivity between the customer premises and Telkom VoIP network that is used to transport the voice call traffic to and from the customer.
2. SIP Sessions – This is number of concurrent calls (incoming/outgoing) that will be allowed over a SIP Trunk.
3. Number range – This is a range of 10-digit numbers that are used to make calls or receive voice calls.
   - » 10 new numbers can be allocated for every 1 SIP session, i.e. 20 SIP sessions will receive a range consisting of 200 numbers.
   - » Keep existing numbers - Existing BRI or PRI number ranges can be ported to a Telkom SIP Trunk.
   - » Single numbers (Junction lines and BRI) can be migrated to a SIP Trunk

# Sip trunking continued…..

- Sip Trunk replaces conventional lines; Junctions/ISDN - **(Cost Savings)**
- An IP PBX that is SIP enabled is required for SIP
- Need broadband access for SIP trunking
- No. of Sip sessions determines the number of concurrent calls
- Customers can migrate their existing numbers to SIP
- Bandwidth determines the maximum number of sessions/concurrent calls (need 40kbps per voice call & 100kbps for a fax call)
- Same connection that the business uses for Internet use can be used for SIP calls - **(Cost Savings)**
- Can get competitive calling rates by simply adding a calling plan (e.g. BizTalk SIP, BizTalk VoIP, VoIP Calling plans) - **(Cost Savings)**
- Customer on no calling plan pays ± R1.14 p/min on standard billing

# Sip Trunking Benefits

- Cost reduction
  - Multiple conventional lines reduced to one single SIP trunk.
  - Reduced call rates on a calling plan
- Flexibility
  - Any number of SIP sessions, ranging from 2 onwards can be provided.
  - Any broadband access can be used (e.g. Fibre, DSL, VDSL)
  - SIP sessions can be easily increased/decreased without penalties
- Business Continuity
  - Existing numbers can be migrated to SIP & Smart access still works on SIP

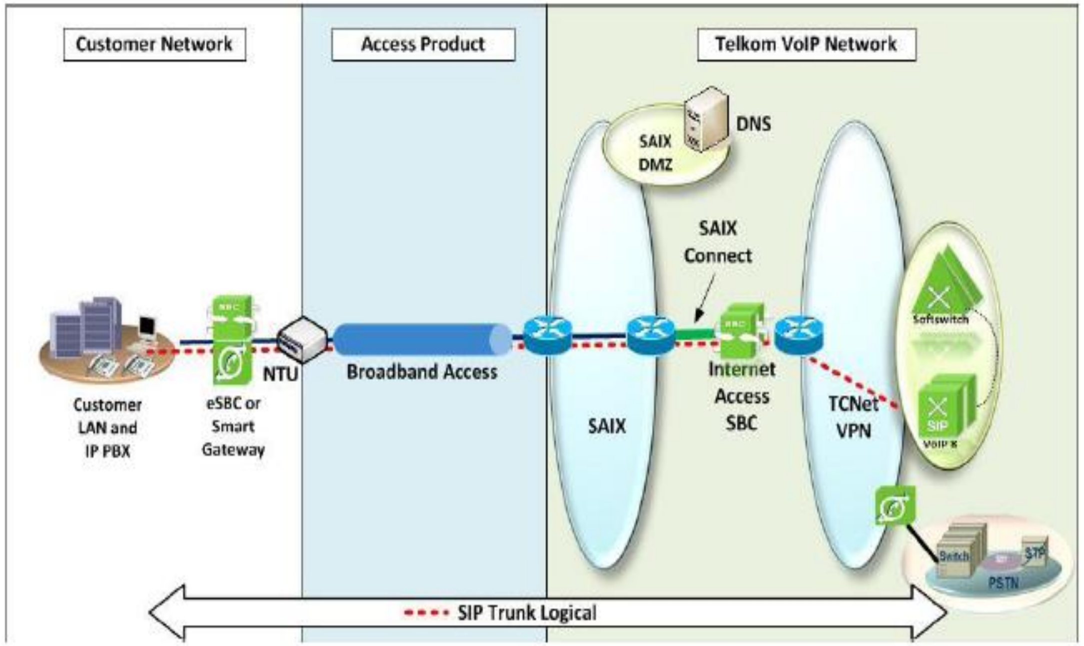
# Standard Sip Deployment

- User name and password
  - A username and password will be provided, per SIP Trunk, to the customer
  - The username and password will allow access to the SIP service (pre-programmed on PBX)

- PBX is required
  - An IP PBX, which is SIP capable, is required to terminate the SIP Trunk.

# 2 Types of SIP Trunking products

## Standard SIP

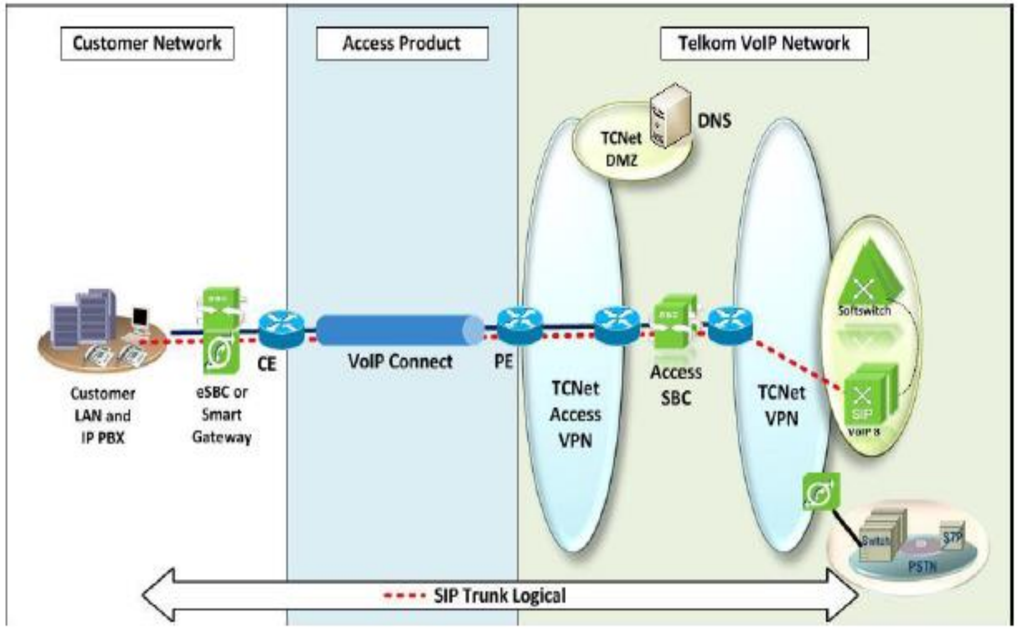



✓ **Standard SIP** uses both broadband (Retail fibre, DSL) & dedicated data

- Targeted at the Large Small & Medium customer
- No QOS on voice
- Suitable replacement for Junctions, BRI and PRI lines

&

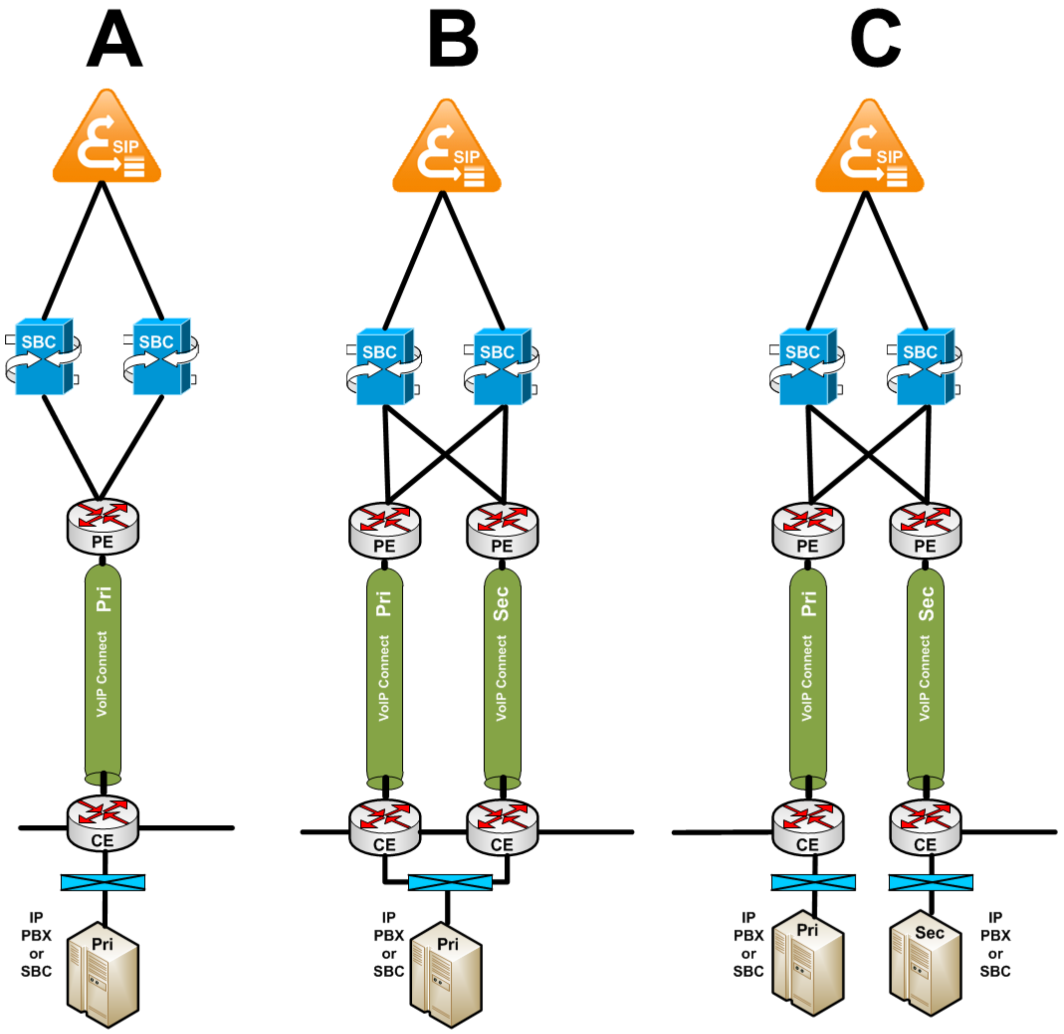
## Premium SIP




✓ **Premium SIP** uses dedicated broadband (e.g. ME)

- Targeted at the Large Enterprise customer
- Voice prioritisation with VoIP connect access
- Redundancy
- SBC (load balancing, QOS marking, security)
- Suitable replacement for PRI lines (base min.of 30 sessions)

# Premium SIP Redundancy options



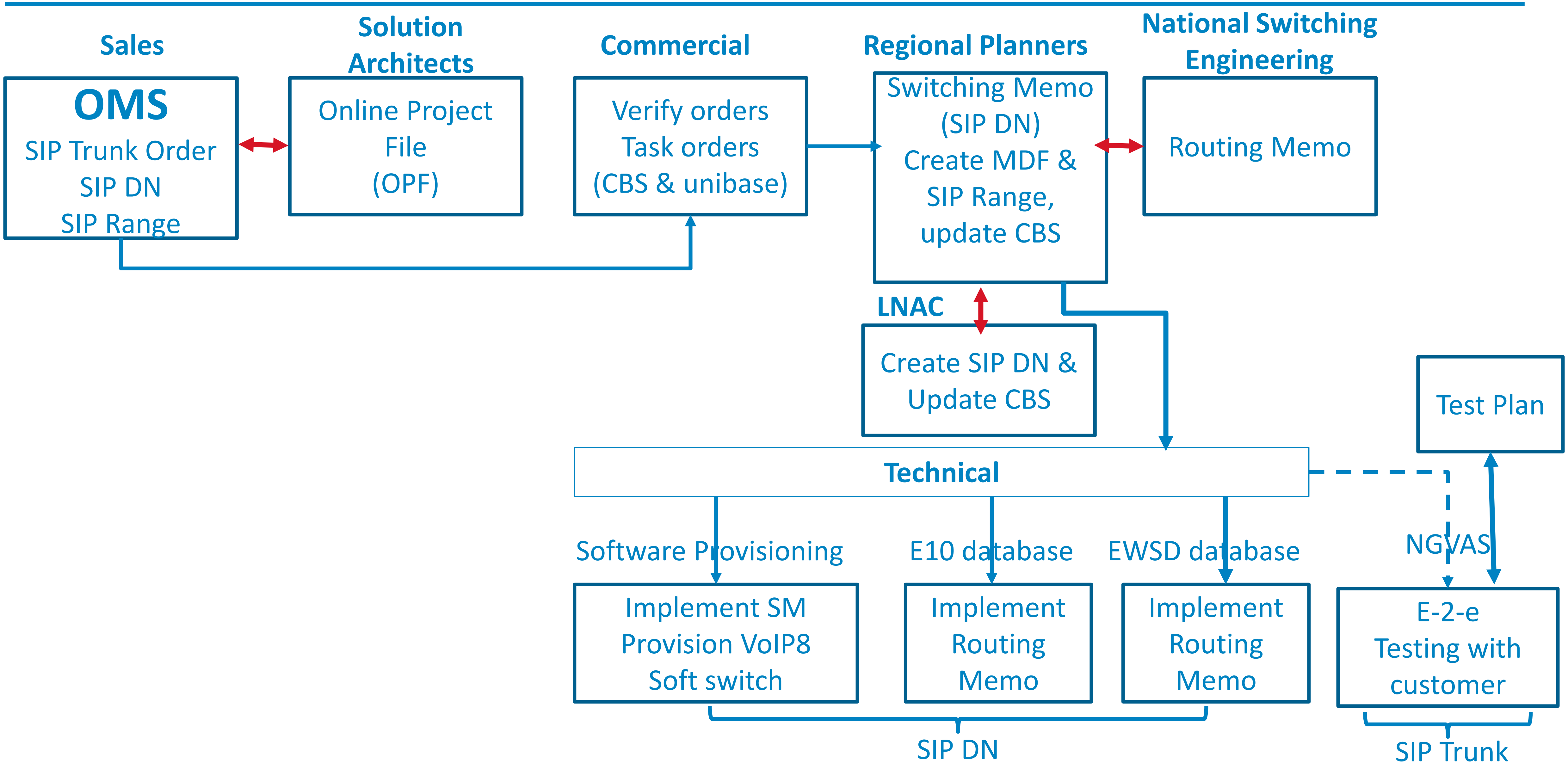

Type A – Single Site, no redundancy

Type B – Single Site, “East-West” building redundancy

Type C – Geographical Redundancy

*Redundancy is offered by load balancing SIP Trunking (Single SIP Trunk) traffic across two VoIP Connects (50/50) in real time. Should a any VoIP Connect access path fail the 100% of traffic will flows down the surviving path. Active calls on the fail link will be dropped and must be re-established over the surviving link.*

# High Level SIP Trunk Provisioning




KEEP
CALM
AND
DRINK
COFFEE
THANK YOU
Telkom