Dial-up Modem migration from PRI to SIP Trunks

Client Requirement Summary

- Maintain dial-up modem service after PRI service is discontinued
- No change to customer equipment
- Direct functional replacement for existing (obsolete) RAS systems
- SIP trunk support
- No End of Sale or End of Life hardware components
- Support for 8+ years

Key Benefits

- SIP trunks to replace PRI lines
- Legacy RAS can be turned off
- No change to customer's existing equipment
- Maintain dial-up service
- Functional replacement for the legacy RAS
- PRI interface
- All common PSTN standards
- ISDN and V.110/V.120
- Fully supported up-to-date solution
- Easy to deploy and manage remote site devices with support for management functions such as RADIUS
- SIP trunking function does not require external clock source.

Requirement

The client provides a dial-up modem service aggregated on PRI lines and terminated by RAS devices. ISDN service will be end of life in the next year and therefore PRI lines will not be available after this date. The client attempted to convert the PRI line to SIP using a media Gateway. However, this 'VoIP in the core' approach results in failed calls due to TDM interfaces not being synchronised resulting in an unreliable service.

An option for the communications provider would be to terminate the dial-service, however, many of their customers have equipment that only supports dial interfaces and upgrading equipment is too costly or disruptive. Customer applications are often in areas such as remote telemetry for water, gas, power, and other high profile activities where minimising disruption is essential. The client needed to implement a reliable service that would allow customers to keep equipment with dial-up interfaces on SIP after PRI service has been discontinued. Any replacement must guarantee availability of hardware and software for 8+ years.

Virtual Access Solution

The Virtual Access RAS SIP trunk functionality is implemented using a unique algorithm that means it is not possible for the IP RAS to lose synchronisation unless the network itself is very poor. This eliminates failed calls caused by TDM interfaces not being synchronised due to the use of 'VoIP in the core'. Support for 10+ years is possible since it is a pure software implementation and does not use aging modem chipsets or DSPs. The SIP trunking function does not require an external clock source and compensates for deviance in timing from independent clocks of multiple clients to the same destination.