# SIP Voice

SIP Voice is available to customers of My Net Pty Ltd via our wholesale arrangement with AAPT. SIP Voice is a SIP Trunking solution. It allows customers with an IP enabled PBX or SIP Gateway device to connect via Ethernet and have their telephony traffic carried via IP utilising Session Initiation Protocol (SIP). It provides a far more scalable alternative to traditional ISDN.

## SIP Voice Specifications

| SERVICE PARAMETER | FEATURES |
|---|---|
| **Service Variants** | 10 SIP Sessions scalable to 300+ SIP Sessions. |
| **Service Availability** | 99.95% |
| **Service Coverage** | The AAPT SIP Voice service can be delivered to any of the following locations served by AAPT Ethernet (including Mid-Band Ethernet): Adelaide, Brisbane, Canberra, Gold Coast, Melbourne, Newcastle, Perth and Sydney.<br>Calls can originate and be terminated anywhere in Australia and abroad from fixed and mobile networks. |
| **Bandwidth Available** | 2Mbps scalable to 30+Mbps. |
| **Interface** | Ethernet 10/100BaseT or Ethernet 1Gbps interface (1000BASE-TX/SX/LX). |
| **Access Method** | AAPT SIP Voice is delivered on the following AAPT Ethernet Physical Access options (and selected third party coverage):<br>• Ethernet Trunk Access (ETA)<br>• Ethernet Single-Service Access (ESSA) for a dedicated voice only solution (viable for 30+ Session Services).• Ethernet Multi-Service Access (EMSA) |
| **Access Redundancy/Overflow** | Optional call failover from SIP Voice to PRI Trunks. |
| **Network Termination Points** | The default electrical interface for an AAPT SIP Voice service is Ethernet. |
| **Conforming Standards** | SIP Voice Trunks are provisioned to conform to RFC3261, using the Session Initiation Protocol (SIP). |
| **Access Configuration** | Both-way: AAPT carries the end customer's outgoing and incoming calls.<br>Outgoing only: AAPT carries outgoing calls, while the customer can retain their incumbent carrier for incoming calls. |
| **Signalling** | Session Initiation Protocol (SIP). |
| **Codec** | G.711 codec – no voice compression. |
| **SIP Trunk Service IP Address Range** | AAPT will allocate customers a /30, /29 or /28 IP Address Range for each individual SIP Voice Access. This enables the following useable IP addresses per option: |
| **Features** | Direct In-Dial, Calling Line Identification (either Calling Line ID Presented (CLIP) or Calling Line ID Restricted (CLIR)), Extension Level Billing or Main Billing Number, Local Number Portability, New AAPT Number Ranges, Number Range Reservation, Directory Listing Facilitation; Optional Call Overflow/Failover to PRI Trunks, Call Redirection and Disaster Recovery Call Redirection. |
| **Originating Calls** | Outgoing voice calling services include Local, National Long Distance, Fixed to Mobile, International Long Distance (fixed line and mobiles), Free Phone (1800), Local Rate (13 and 1300), Directory Services, a selected list of Information Service numbers (190X), Emergency Services, Selected Special Services offered by other carriers are also available. |
| **Terminating Calls** | Incoming voice calling services include Local, National Long Distance, Fixed to Mobile, International Long Distance (fixed line and mobiles), Free Phone (1800), Local Rate services (where the customer has designated an AAPT SIP Voice service to terminate calls made to a 1800 number or 13/1300 number respectively). |
| **Provisioning and Outage Restoration Targets** | Access Install and Physical Modifications (over AAPT infrastructure): Metro: 20 business days (over new<br>Ethernet Access), 10 business days (over existing Ethernet Access). Regional: 30 business days (over new Ethernet Access), 15 business days (over existing Ethernet Access). Logical Modifications (over AAPT infrastructure): 5 business days. Mean Time to Repair (Metro) for AAPT Infrastructure: 4 hours. |