COMMUNICATIONS

ALLIANCE LTD

INDUSTRY GUIDELINE

G672:2023

SESSION INITIATION PROTOCOL (SIP) INTERCONNECTION

G672:2023 Session Initiation Protocol (SIP) Interconnection Industry Guideline

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Communications Alliance Ltd was formed in 1997 to provide a unified voice for the Australian communications industry and to lead it into the next generation of converging networks, technologies and services.

INTRODUCTORY STATEMENT

The ***Session Initiation Protocol (SIP) Interconnection*** Guideline (G672:2023) is designed to:

- Assist a Carrier or Carriage Service Provider (CSP) seeking interconnection of services via Australian Telecommunications Networks;
- Guide on what to expect when implementing interconnection; and
- Help maintain the end-to-end integrity of Telecommunications Networks.

James Duck
Chair
SIP Interconnect *Working Committee*

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TABLE OF CONTENTS

1 GENERAL

1.1 Introduction

1.1.1 The Guideline is an outcome from effort by the Australian telecommunications industry to identify generally agreed elements a Carrier or Carriage Service Provider (CSP) should consider for inclusion in bilateral agreements for the interconnection of Telecommunications Networks and services that use the Session Initiation Protocol (SIP).

1.1.2 Mobile and fixed networks have similar but different starting points. Mobile networks typically align with 3GPP specifications, as these are written for mobile networks. Fixed networks usually align with IETF RFCs.

1.1.3 Although there is liaison and cross representation between the relevant IETF and 3GPP committees, there may be differences between the two sets of documentation.

1.2 Scope

1.2.1 The Guideline applies to the following sections of the telecommunications industry under section 110 of the Act:

(a) Carriers; and

(b) CSPs.

1.2.2 It deals with the following telecommunications activities as defined in section 109 of the Act:

(a) carrying on business as a Carrier; or

(b) supplying goods or service(s) for use in connection with the supply of a Listed Carriage Service.

1.2.3 The Guideline deals with the interconnection of calls at the Network-Network Interface that:

(a) use the Session Initiation Protocol (SIP);

(b) are inside Australia; and

(c) originate from Customer Equipment (CE).

1.2.4 The Guideline does not deal with the interconnection of calls that use:

(a) Signalling System No.7; or

(b) ISDN User Part.

1.2.5 The Guideline does not deal with the interconnection of calls at the User-Network Interface.

1.3 Objectives

The objectives of the Guideline are to:

(a) Assist a Carrier or CSP seeking interconnection of services with Australian Telecommunications Networks;

(b) Guide on what to expect when implementing interconnection; and

(c) Help maintain the end-to-end integrity of Telecommunications Network.

1.4 Guideline review

The Guideline will be reviewed every 5 years, or earlier in the event of significant developments that affect the Guideline.

2 ACRONYMS, DEFINITIONS AND INTERPRETATIONS

2.1 Acronyms

For the purposes of the Guideline:

3GPP

3rd Generation Partnership Program.

AMR-WB

Adaptive Multi-Rate Wideband.

CLI

Calling Line Identification.

CSP

Carriage Service Provider

IETF RFC

Internet Engineering Task Force Request for Comment.

IP

Internet Protocol.

ITU

International Telecommunications Union.

ITU-T

ITU Telecommunications standardisation sector.

MoLI

Mobile Location Information.

NNI

Network-Network Interface.

PANI

P Access Network Identifier.

PIDF

Presence Information Data Format.

PIDF-LO

Presence Information Data Format Location Object.

RTP

Real-time Transport Protocol

SDP

Session Description Protocol

SIP

Session Initiation Protocol

TCP

Transmission Control Protocol

UDP

User Datagram Protocol

2.2 Definitions

For the purposes of the Guideline:

Act

means the *Telecommunications Act 1997 (Cth)*.

Carriage Service Provider

has the meaning given by section 87 of the Act.

Carrier

has the meaning given by section 7 of the Act.

Customer Equipment

has the meaning given by section 21 of the Act.

Determination

means the *Telecommunications (Emergency Call Service) Determination 2019*.

Emergency Call

has the meaning given by the Determination.

Emergency Service Number

has the meaning given by section 30 of the *Telecommunications Numbering Plan 2015*.

Header Field

has the meaning given by IETF RFC 3261.

Session Initiation Protocol (SIP)

has the meaning given by IETF RFC 3261.

Telecommunications Network

has the meaning given by section 7 of the Act**.**

2.3 Interpretations

In the Code, unless the contrary appears:

(a) headings are for convenience only and do not affect interpretation;

(b) a reference to a statute, ordinance, code or other law includes regulations and other instruments under it and consolidations, amendments, re-enactments or replacements of any of them;

(c) words in the singular includes the plural and vice versa;

(d) words importing persons include a body whether corporate, politic or otherwise;

(e) where a word or phrase is defined, its other grammatical forms have a corresponding meaning;

(f) mentioning anything after include, includes or including does not limit what else might be included;

(g) words and expressions which are not defined have the meanings given to them in the Act; and

(h) a reference to a person includes a reference to the person's executors, administrators, successors, agents, assignees and novatees.

3 REQUIREMENTS

3.1 SIP

Refer to Table 1 for supported technical specifications.

TABLE 1
Technical Specification Support

Description	Reference / Source Document	Status	Details / Comments
Transport			
TCP: Transmission Control Protocol	IETF RFC 9293	Preferred	The use of TCP for SIP transport across the NNI is preferred ahead of using UDP
UDP: User Datagram Protocol	IETF RFC 768	Supported	
SIP			
SIP: Session Initiation Protocol	IETF RFC 3261	Supported	
SDP: An Offer/Answer Model with SDP	IETF RFC 3264	Supported	
RTP: A Transport Protocol for Real-Time Applications	IETF RFC 3550	Supported	
PRACK: Reliability of provisional responses in the SIP	IETF RFC 3262	Supported	
UPDATE: SIP Update Method	IETF RFC 3311	Supported	
REASON: The Reason Header Field for the SIP	IETF RFC 3326	Supported	
Session Timers in SIP	IETF RFC 4028	Supported	
Codec			
G711a	ITU-T Rec. G.711	Supported	G711a is the default narrowband codec in Australia for call origination on fixed networks
AMR		Supported	AMR (narrowband) is the default codec in Australia for call origination on mobile networks
G722.2	ITU-T Rec. G.722.2 / IETF RFC 4867	Supported	Also known as AMR-WB
G.729	ITU-T Rec. G.729	Supported	For backwards compatibility

Description	Reference / Source Document	Status	Details / Comments
Dual Tone Multi-Frequency (DTMF)			
RTP Payload for DTMF digits	IETF RFC 2833 IETF RFC 4733	Supported	Preferred ahead of in band transmission of DTMF tones.
DTMF In-band via G711		Supported	For backwards compatibility
Call Hold			
An Offer/Answer Model with SDP	IETF RFC 3264	Supported	
Call Diversion			
History Info Header	IETF RFC 7044	Supported	Mandatory for call diversion
Diversion Header	IETF RFC 5806	Supported	Redundant if using History Info header
URI Format			
SIP URI	IETF RFC 3261	Supported	
Tel URI	IETF RFC 3966	Supported	
Call screening, Privacy, CLI Presentation			
A privacy Mechanism for SIP	IETF RFC 3323	Supported	
P-Asserted-Identity Header	IETF RFC 3325	Supported	
Charging			
P-Charging Vector	IETF RFC 7315	Supported	May be required for IMS based networks
IP Quality of Service (DSCP)			
Call signalling (CS5) for SIP	IETF RFC 4594		SIP DSCP 40/CS5
Expedited Forwarding (EF) for RTP	IETF RFC 3246		Voice DSCP 46/EF
Number Format			
E.164	ITU-T Rec. E.164	Supported	

3.2 Alignment with Standards

Information forwarded in SIP Header Fields by a Carrier should align with relevant standards (e.g. IETF RFCs, 3GPP TSs).

3.3 Codecs and Transcoding

3.3.1 A list of supported codecs is subject to bilateral negotiation.

3.3.2 A list of codecs in a SIP offer should include:

(a) Multiple codecs; and

(b) The default codec.

3.3.3 In Australia the default codec for originating calls from a:

(a) Fixed network is G.711a; and

(b) Mobile network is AMR.

3.3.4 Use of a wideband codec in a call will depend on its support in both user endpoints.

3.3.5 Rules for transcoding are subject to bilateral negotiation.

3.3.6 A guiding principle is to minimise the use of transcoding.

3.3.7 The content of a mid-call offer is subject to bilateral negotiation.

3.3.8 A mid-call offer should, at a minimum, offer the previously answered codec.

3.4 Call Hold

Call hold agreements are subject to bilateral negotiation.

3.5 Call Diversion

3.5.1 A SIP header for interconnecting diverted calls should include forwarding information.

3.5.2 History Info (refer to RFC 7044) should be the default SIP header for providing forwarding information about diverted calls.

3.6 Call screening, Privacy, CLI Presentation

3.6.1 A Carrier or CSP is required to provide an interconnecting Carrier or CSP with CLI information for a call.

> *NOTE: Providing CLI information may enable a Carrier or CSP to:*
>
> *(a) bill a Customer;*
>
> *(b) route a Communication;*
>
> *(c) verify the location of a Calling Party;*
>
> *(d) validate that the Calling Party is entitled to use the Carriage Service or Content Service;*
>
> *(e) perform lawful inbound call centre number display applications to improve the quality of service provided by those call centres;*
>
> *(f) use CLI information as an input for a party to comply with the obligations in C661;*

(g) use CLI information in a manner otherwise required or permitted by law; or

(h) use CLI information for other purposes (e.g. calling number display by using CLI Presentation details) as agreed bilaterally by the Carrier(s) or CSP(s).

3.6.2 The delivery of CLI information is subject to bilateral negotiation e.g. via P-Asserted-Identity Header (refer to RFC 3325).

3.6.3 Where a CLI for the caller is not available (e.g. some Emergency Calls) the originating Carrier should include at least one default CLI. Multiple CLIs may be included as well.

3.6.4 Arrangements that define default CLI(s) and their use are subject to bilateral negotiation.

3.6.5 The applicable headers, P-Asserted-Identity and From containing these CLIs should be formatted as a SIP URI, with the host field identifying the Fully Qualified Domain Name (FQDN) of the Interconnecting Carrier.

3.7 In Band Data Communications

3.7.1 Calls from existing analogue customer terminals that use in-band data communications (e.g. fax, EFTPOS terminals, text telephones devices (TTY) and back-to-base alarm terminals) may not be compatible with IP networks.

3.7.2 Support of in-band data communications (e.g. fax, based on IETF RFC 3362 / ITU-T Rec. T.38) is:

(a) subject to bilateral negotiation; and

(b) likely to be on a 'best efforts' basis.

NOTE: Fax is not supported in all networks.

3.8 IP Quality of Service (DSCP)

3.8.1 A Carrier or CSP should use the Differentiated Services Control Protocol (DSCP) to signal Quality of Service (QoS) across the NNI.

3.8.2 A Carrier or CSP should support the IP Type-of-Service (DSCP) QoS markings.

NOTES:

1. The interconnected networks should be capacity managed for no packet loss.

2. All bearers should be bi-directional and symmetric with respect to attributes e.g. bit rate and metrics.

3. For more information on Quality of Service (QoS) parameters for IP networks refer to G632.

> *4. For more information on Quality of Service (QoS) parameters for Voice over Internet Protocol (VoIP) services refer to G634.*

3.9 Number Format

3.9.1 All phone numbers should use the global format (refer to RFC 3966).

> *NOTES:*
>
> *1. SIP requires the E.164 number format.*
>
> *2. The Telecommunications Numbering Plan 2015 specifies some numbers that are not E.164 numbers. For example, it defines 000 and 106 as Emergency Service Numbers.*

3.9.2 SIP URIs should carry the 'user=phone' URI parameter.

3.9.3 For either SIP or tel URIs all routable numbers should be formatted as a global-number with the digits in E.164 international format.

> *NOTES:*
>
> *1. For example:*
>
> *(a) sip:+614xxxxxxxx;user=phone...*
>
> *(b) tel:+614xxxxxxxx*
>
> *2. The support of any number format other than E.164 (e.g. a national significant number) is subject to bilateral negotiation.*

3.9.4 All global service numbers, including Inbound 180X/13/1300 numbers should be passed in global-number format.

> *NOTE: This means location information (e.g. MoLI) is carried in a SIP Header Field and is not embedded within the called number digits string (B-number).*

3.9.5 All numbering information should be carried enbloc only.

3.10 Location Information

3.10.1 The delivery of location information is subject to bilateral negotiation.

3.10.2 Location information should be carried in the SIP P Access Network-Info header.

> *NOTES:*
>
> *1. For example: P-Access-Network-Info: GSTN;operator-specific-GI="ABC";network-provided*

2. This form of P-Access-Network-Info header is referred to as "PANI-GI". The double-quotes are literals and are required to envelope the ABC value for location information.

3. Carriage of ABC MoLI (refer to G557.2 and G557.5) in PANI-GI is expected to be required indefinitely.

4. Possible future enhancements for location information (for further study) include SIP Geolocation headers (refer to RFC 6442) and PIDF-LO (refer to RFC 4119) message-body content.

5. A Presence Information Document Format – Location Object (PIDF-LO) provides detailed location information formatted as an XML document and carried as MIME content within the body of a SIP request. The Geolocation header identifies the location as a URI which acts as a pointer to the PIDF-LO content in the SIP message body.

3.10.3 Location information should not be embedded within the called number digits string (B-number) of the INVITE Request Line.

3.11 Packetization

The recommended ptime value is 20ms.

NOTES

1. *A ptime of 10ms may also be supported by some Carriers.*
2. *Refer to G.634 for more information on the bandwidth impacts of different voice sample sizes for G.711 and G.729 codecs.*

3.12 Call Authentication

Call authentication may be negotiated on a bilateral basis in accordance with C661.

3.13 Failover

Arrangements between Carrier for failover are the subject of bilateral negotiation.

NOTES:

1. This allows Carriers to put in place failover arrangements that are subject to carrier specific matters e.g. individual set up of Carrier equipment.

2. A possible example is a call set up does not receive a code 200 (OK) within the expected time then the failover arrangements commence, with 503 (Service Unavailable) a popular choice for a response code.

3.14 SIP OPTIONs

The support of OPTIONs is the subject of bilateral negotiation, being dependent on carrier equipment.

3.15 Interconnect Service Information

The passing of interconnect service information across the SIP Interconnect NNI is subject to bilateral negotiation.

3.16 Routing of (ported) numbers

3.16.1 A Carrier originating a call to a number that may be subject to number portability (e.g. freephone, local rate, geographic and mobile numbers) should check the ported number status of the called number or associated number block prior to initiating the call across a SIP POI.

3.16.2 A Carrier should determine the network where a number associated with a service resides before attempting to route traffic towards a destination network.

3.16.3 Support of legacy interconnection arrangements (e.g. G549) is out of scope for this document and a matter for separate bilateral negotiation.

4 SIP RESPONSE CODES

4.1 SIP 1XX (Provisional) Response Codes

Refer to Table 2 for a list of industry supported SIP 1XX (Provisional) codes and related comments.

TABLE 2

Industry supported SIP 1XX (Provisional) response codes

SIP Request Code	Code description	Comments
100	Trying	
180	Ringing	
181	Call Is Being Forwarded	
182	Queued	
183	Session Progress	
199	Early Dialog Terminated	

NOTES:

1. Refer to Section 21.1 of IETF RFC3261 for the definition of 1XX (Provisional) response codes.

2. Refer to IETF RFC6228 for the definition of Provisional response code 199 (Early Dialog Terminated).

4.2 SIP 2XX (Successful) Response Codes

Refer to Table 3 for a list of industry supported SIP 2XX (Successful) response codes and related comments.

TABLE 3

Industry supported SIP 2XX (Successful) response codes

SIP Request Code	Code description	Comments
200	OK	
204	No Notification	

NOTE: Refer to Section 21.2 of IETF RFC3261 for the definition of 2XX (Successful) response codes.

4.3 SIP 3XX (Redirection) Response Codes – Not Supported

Refer to Table 4 for a list of SIP 3XX (Redirection) response codes that are not supported at points of interconnection in the Australian telecommunications industry, and related comments.

TABLE 4

Industry excluded SIP 3XX (Redirection) response codes

SIP Request Code	Code description	Comments
300	Multiple Choices	
301	Moved Permanently	
302	Moved Temporarily	
305	Use Proxy	
380	Alternative Service	

NOTE: Refer to Section 21.3 of IETF RFC3261 for the definition of 3XX (Redirection) response codes.

4.4 SIP 4XX (Request Failure) Response Codes

Refer to Table 5 for a list of industry supported SIP 4XX (Request Failure) codes and related comments.

TABLE 5

Industry supported SIP 4XX (Request Failure) response codes

SIP Request Failure Code	Code description	Comments
400	Bad Request	
403	Forbidden	
404	Not Found	Do not retry through other handovers.
405	Method Not Allowed	
406	Not Acceptable	
407	Proxy Authentication Required	
408	Request Timeout	
410	Gone	
413	Request Entity Too Large	
414	Request-URI Too Long	
415	Unsupported Media Type	
416	Unsupported URI Scheme	
420	Bad Extension	
421	Extension Required	
423	Interval Too Brief	
440	Max-Breadth Exceeded	
480	Temporarily Unavailable	Do not retry through other handovers.

481	Call/Transaction Does Not Exist	
482	Loop Detected	
483	Too Many Hops	
484	Address Incomplete	
485	Ambiguous	
486	Busy Here	
487	Request Terminated	
488	Not Acceptable Here	
491	Request Pending	
493	Undecipherable	

NOTE: Refer to Section 21.4 of IETF RFC3261 for the definition of 4XX (Request Failure) responses.

4.5 SIP 5XX (Server Failure) Response Codes

Refer to Table 6 for a list of industry supported SIP 5XX (Server Failure) response codes and related comments.

TABLE 6

Industry supported SIP 5XX (Server Failure) response codes

SIP Request Failure Code	Code description	Comments
500	Server Internal Error	
501	Not Implemented	
502	Bad Gateway	
503	Unavailable	SIPUA is offline if response not received.
504	Server Time-out	
505	Version Not Supported	
513	Message Too Large	
580	Precondition Failure	

NOTES:

1. Refer to Section 21.5 of IETF RFC3261 for the definition of most 5XX (Server Failure) response codes.

2. Refer to Section 8 of IETF RFC3312 for the definition of Server Failure response code 580 (Precondition Failure).

4.6 SIP 6XX (Global Failure) Response Codes

Refer to Table 7 for a list of industry supported SIP 6XX (Global Failure) response codes and related comments.

TABLE 7

Industry supported SIP 6XX (Global Failure) response codes

SIP Request Failure Code	Code description	Comments
600	Busy Everywhere	
603	Decline	
604	Does Not Exist Anywhere	
606	Not Acceptable	

NOTE: Refer to Section 21.6 of IETF RFC3261 for the definition of 6XX (Global Failure) response codes.

5 SIP PARAMETERS

5.1 Call Control

The following SIP parameters should apply:

(a) Early and mid-call dialog UPDATE(s) shall be supported.

(b) Receipt of SIP URI with “user=phone” scheme is recommended.

(c) Request URI the tel URI (RFC 3966) scheme shall also be supported.

(d) Forwarded calls should be indicated via SIP History-Info header.

(e) Mid Call Codec Negotiation/ Modification using re-INVITE should be supported.

(f) Delayed offer, initial INVITE and mid-call re-INVITE should be supported.

6 REFERENCES

Publication	Title
Industry Codes	
C661:2022	Reducing Scam Calls and Scam SMs https://www.commsalliance.com.au/Documents/all/codes/c661
Industry Guidelines	
	Location Information for Emergency Calls
G557.2:2014	Part 2: Standardised Mobile Service Area (SMSA) and Location Indicator Register
G557.5:2021	Part 5: Push Mobile Location Information (MoLI) Interface To Emergency Call Person Platform (ECPP) https://www.commsalliance.com.au/Documents/all/guidelines/g557
G632:2012	Quality of Service parameters for networks using the Internet Protocol https://www.commsalliance.com.au/Documents/all/guidelines/g632
G634:2013	Quality of Service parameters for Voice over Internet Protocol (VoIP) services https://www.commsalliance.com.au/Documents/all/guidelines/g634
IETF RFCs	
RFC 768	User Datagram Protocol https://www.rfc-editor.org/info/rfc768
RFC 3246	An Expedited Forwarding PHB (Per-Hop Behavior) https://www.rfc-editor.org/info/rfc3246
RFC 3261	SIP: Session Initiation Protocol https://www.rfc-editor.org/info/rfc3261
RFC 3262	PRACK: Reliability of provisional responses in the SIP https://www.rfc-editor.org/info/rfc3262
RFC 3264	An Offer/Answer Model with Session Description Protocol (SDP) https://www.rfc-editor.org/info/rfc3264

RFC 3311	UPDATE: SIP Update Method https://www.rfc-editor.org/info/rfc3311
RFC 3312	Integration of Resource Management and Session Initiation Protocol (SIP) https://www.rfc-editor.org/info/rfc3312
RFC 3325	Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks https://www.rfc-editor.org/info/rfc3325
RFC 3326	REASON: The Reason Header Field for the SIP https://www.rfc-editor.org/info/rfc3326
RFC 3362	Real-time Facsimile (T.38) - image/t38 MIME Sub-type Registration https://www.rfc-editor.org/info/rfc3362
RFC 3550	RTP: A Transport Protocol for Real-Time Applications https://www.rfc-editor.org/info/rfc3550
RFC 3966	The tel URI for Telephone Numbers https://www.rfc-editor.org/info/rfc3966
RFC 4028	Session Timers in SIP https://www.rfc-editor.org/info/rfc4028
RFC 4119	A Presence-based GEOPRIV Location Object Format https://www.rfc-editor.org/info/rfc4119
RFC 4594	Configuration Guidelines for DiffServ Service Classes https://www.rfc-editor.org/info/rfc4594
RFC4733	RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals https://www.rfc-editor.org/info/rfc4733
RFC 4867	RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs https://www.rfc-editor.org/info/rfc4867
RFC 5031	A Uniform Resource Name (URN) for Emergency and Other Well-Known Services https://www.rfc-editor.org/info/rfc5031

RFC 5806	Diversion Indication in SIP https://www.rfc-editor.org/info/rfc5806
RFC 6228	Session Initiation Protocol (SIP) Response Code for Indication of Terminated Dialog https://www.rfc-editor.org/info/rfc6228
RFC 7044	An Extension to the Session Initiation Protocol (SIP) for Request History Information https://www.rfc-editor.org/info/rfc7044
RFC 7315	Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3GPP https://www.rfc-editor.org/info/rfc7315
ITU-T Recommendations	
E.164 (11/2010)	The international public telecommunication numbering plan https://www.itu.int/itu-t/recommendations/rec.aspx?rec=10688
G.711 (11/88)	G.711 : Pulse code modulation (PCM) of voice frequencies https://www.itu.int/rec/T-REC-G.711/en
G.722.2 (07/03)	G.722.2 : Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB) https://www.itu.int/rec/T-REC-G.722.2/en
G.729 (06/12)	G.729 : Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP) https://www.itu.int/rec/T-REC-G.729/en
T.38 (11/2015)	Procedures for real-time Group 3 facsimile communication over IP networks https://www.itu.int/rec/T-REC-T.38/en
Legislation	
Telecommunications Act 1997 *https://www.legislation.gov.au/Series/C2004A05145*	

Telecommunications (Emergency Call Service) Determination 2019 *https://www.legislation.gov.au/Series/F2019L01509*	
Telecommunications Numbering Plan 2015 *https://www.legislation.gov.au/Series/F2015L00319*	
Other	
-	ACMA Register of Other Numbers https://www.acma.gov.au/publications/2019-11/data/register-other-numbers

PARTICIPANTS

The Working Committee that developed the Guideline consisted of the following organisations and their representatives:

The Working Committee responsible for the revisions made to this Code consisted of the following organisations and their representatives:

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This Working Committee was chaired by James Duck of Communications Alliance who also provided project management support.

Communications Alliance was formed in 1997 to provide a unified voice for the Australian communications industry and to lead it into the next generation of converging networks, technologies and services.

In pursuing its goals, Communications Alliance offers a forum for the industry to make coherent and constructive contributions to policy development and debate.

Communications Alliance seeks to facilitate open, effective and ethical competition between service providers while ensuring efficient, safe operation of networks, the provision of innovative services and the enhancement of consumer outcomes.

It is committed to the achievement of the policy objective of the *Telecommunications Act 1997* - the greatest practicable use of industry self-regulation without imposing undue financial and administrative burdens on industry.

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