



"Offering **integrated**
Voice,
Web
& Mobile
Solutions **&**
Services
which
make
communication
more **effective**
& more
efficient"



Comsys SpeechFrame

Services & Solutions for

COMSYS

Voice
Web
Mobile

Reference number: JST.SFO.GNRL (SFO-03)
Soesterberg, June 2009

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Document management

Version	Date	Description	Author
0.1	03-07-2008	First concept	Joost van der Stelt, Comsys Telecom & Media B.V.
0.2	03-10-2008	First draft	Joost van der Stelt, Comsys Telecom & Media B.V.
1.0	03-11-2008	First release	Joost van der Stelt, Comsys Telecom & Media B.V.
1.1	06-25-2009	Updated to reflect current release	Joost van der Stelt, Comsys Telecom & Media B.V.

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1. Introduction

The Comsys SpeechFrame platform is based on Java, CCXML and VoiceXML. The platform is fully redundant, highly scalable with guaranteed performance as expected from Telco Grade solutions.

1.1. Platform applications

Platform applications can be written in Java as a 'in the box' solution or in CCXML/VoiceXML. The VoiceXML markup language pages are serviced by a web server. Using VoiceXML introduces flexibility and extreme reduced application development time as the technique is utmost flexible and easy to implement. Time to market for new applications and for desired changes is extremely short.

The CCXML markup language provides Call Control capabilities based on events. Actions to be taken when receiving telephony network events can be defined. The SpeechFrame CCXML processor allows events to be send from external applications towards the processor and the CCXML processor can re-distribute incoming events towards external applications. This allows full control over the platform based on external rules.

1.2. Service creation environment

Next to the development in VoiceXML, Comsys also has a graphical, web based, VoiceXML editor in its portfolio. The editor is easy to understand and can especially be used for end-users who have no experience building web pages whatsoever. For example, call center supervisors can easily add or change dialogues by means of a drag and drop interface. Simply by starting their internet explorer and entering the correct web address.

1.3. Integration

Integration to back-end systems (i.e. databases, billing systems, CRM) is easy because Comsys always deploys Service Oriented Architecture (SOA). From any dialogue, back-end systems can be reached via simple XML data transfer using local or remote web services. Each SOA service can be re-used for other dialogues or even for (existing) internet HTML sites as HTML in essence utilizes the same techniques as CCXML and VoiceXML.

1.4. Operations, Administration and Management

For OA&M, several web based user interfaces are at your disposal. Changing dialogue and system parameters can be done, again, by simply using any internet explorer. The user interfaces are secured by user name and password. Each user can be assigned different roles depending on experience and skills.

All user interfaces use XML data communication to and from the SpeechFrame platform. Integrating external OA&M tools with the SpeechFrame platform is possible based on open standard techniques and protocols.

1.5. Statistics

Statistical information can be retrieved via a web user interface. The information contains call detail records (CDRs) and event detail records (EDRs). The CDRs contain information like start, end and duration of the call. The EDRs contain detailed information of the usage of the dialogue. EDRs can be defined for actions like entering customer numbers, serial numbers and so on. Analyzing the information is extremely useful for optimizing dialogues. All information is direct available and can be exported to numerous external formats, like CSV text files and Excel spreadsheets.

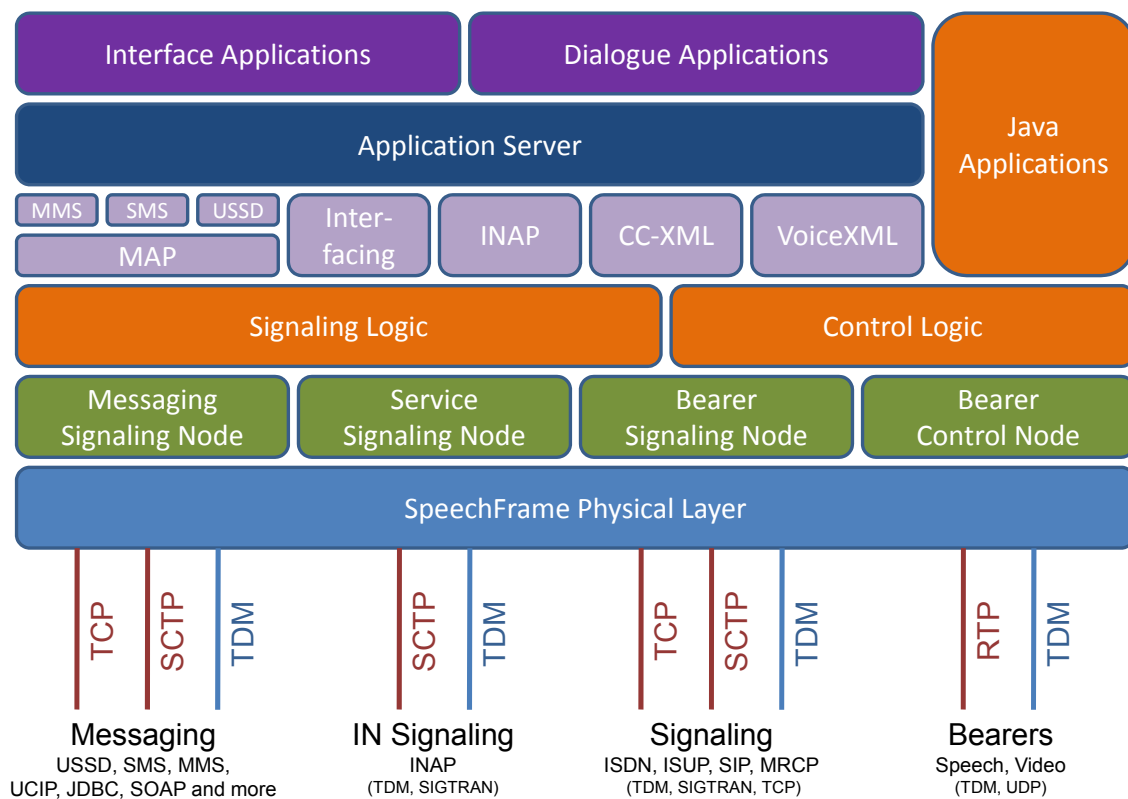
1.6. Monitoring

The system uses SNMP for monitoring hardware, system software, dialogue experience and back-end interfaces processes. On all levels SNMP traps can be defined and sent to centralized operations control. Faulty hardware can be monitored so corrective action can be taken before noticeable problems occur. System software can be monitored in order to keep track on correct functioning of the solution. Traps can be sent from the dialogues at any desired point to monitor, for instance, timing problems towards back-end systems. Monitoring the back-end interface processes allows you to take corrective action immediately if any (software or network) errors occur that can disturb the dialogue from functioning correctly.

2. Architecture

Comsys SpeechFrame allows you to implement the system as stand-alone CCXML/VoiceXML node up to a fully integrated Service Resource Function (SRF) node. The modular architecture is flexible and robust.

2.1. Overview



2.2. Physical Layer

The SpeechFrame physical layer (SPL) consists of platform Ethernet connections and telephony board TDM and/or Ethernet connections.

Network bearers – Speech and Video – can be transported over the TDM connections. The TDM connectors are located on the telephony board in the chassis. Most common bearers are the speech or video channels related to ISDN and ISUP calls. Next to TDM, the bearers can be transported over UDP (RTP). These Ethernet connectors are located on the telephony board in the chassis. Most common are RTP streams related to SIP calls.

2.3. Control Nodes & Signaling Nodes

The control nodes and signaling nodes use 3rd party API's control the device drivers for the SpeechFrame Physical Layer. The nodes are low-level processes, responsible for specific tasks.

2.3.1. Messaging Control Node

The Messaging Control Node (MCN) is responsible for handling non-call related messaging. Messages vary from SMS/MMS/USSD over TDM or SIGTRAN and SMS/MMS/USSD over SMPP/TCP network messaging to dialogue/service specific messaging. These types of messaging are connections to databases (i.e. JBDC), billing systems (i.e. TailorMade MD IVR Interface) and different kind of back-end interfaces based on SOAP/WSDL (i.e. Ericsson UCIP, SchlumbergerSema COSI).

2.3.2. Service Signaling Node

The Service Signaling Node (SSN) is responsible for IN Signaling based on INAP (CS1/CH-IN2). While CS1+ and CS2 are not fully implemented, the SSN has several messages from CS1+ and CS2 implemented. The message set will be expanded on request.

2.3.3. Bearer Signaling Node

The Bearer Signaling Node (BSN) is responsible for all call-related signaling. It handles call control and call switching for the SpeechFrame platform. The BSN utilizes standard protocol signaling and is equipped to handle special signaling scenarios like, for example, ISUP call transfer based on special release cause and ISDN Explicit Call Transfer (ECT) route optimization & on-hold feature.

SpeechFrame can be installed with Euro-ISDN (ETS300-102), ETSI-ISUP (ETS300-356) network protocols. For Voice over IP (VoIP), SIP is available and Media Resource Control Protocol (MRCP) is available for Speech Recognition and Text-To-Speech. Next to these protocols, numerous others can be delivered. For an extensive list, please refer to APPENDIX A.

2.3.4. Bearer Control Node

The Bearer Control Node (BCN) is responsible for media-related control in the SpeechFrame platform. The node handles media resource allocation for playing, recording, conferencing, text-to-speech (TTS) and any other media demand. Besides media control, the BCN also controls bearer channel switching.

2.4. Signaling Logic & Control Logic

The SpeechFrame Signaling Logic (SSL) and SpeechFrame Control Logic (SCL) use Comsys proprietary TCP protocols to communicate with and control the MSN, SSN, BSN and BCN. The units exchange information on the same level based on incoming events and outgoing commands.

Both the SSL and the SCL service the above building blocks, and are in fact the heart of the SpeechFrame system, bridging the gap between low-level nodes and higher-level interpreters or java application.

2.5. Higher level building blocks

These building blocks provide an ease-of-use interface for controlling the SpeechFrame platform. The VoiceXML building block utilizes a VoiceXML 2.0 interpreter for flexible dialogue creation. CCXML is incorporated for call control based on CCXML 1.0.

Next to these two industry standards, building blocks are available for INAP, MMS, SMS and USSD. Several different interfacing building blocks are available, or can be created, to support any back-end interfacing.

Java applications can be written at this level to bypass any restrictions that may be encountered using one of the standards. Also, for extreme mass-calling platforms where handling CCXML and interpreting VoiceXML needs to be excluded, dedicated Java applications can be used.

2.6. Application Server

The application server hosts the (CCXML/VoiceXML) dialogues for the platform. Application servers can run on separate systems, which render the system highly scalable. This will be explained in detail in chapter 3.

2.7. Applications

In effect, the dialogue applications are the services the system hosts. Running on the application server, the dialogue applications, written in VoiceXML, provide the logic for the platform. In the VoiceXML language it is defined when audio/video fragments are played and/or recorded. Transfers can be made to a help desk and back-end systems can be queried.

Interface applications are installed to provide web interfaces, based on XML, for retrieving information. VoiceXML dialogues and user interfaces can use these interfaces to retrieve information that need to be played back to the callers or shown to administrators. Vice versa, information retrieved from caller input (DTMF or Voice Recognition) and data entered in the user interfaces can be inserted or updated in the back-end systems.

For all interfaces needed in dialogue and user interfaces, Comsys creates a SOA data interface. As a result, the Interface Applications can be called from other systems (i.e. web servers) and hereby re-used for other purposes.

3. Capacity & Scalability

Solutions can be deployed starting from a single chassis using a single PCI telephony board (boxed solution), up to horizontally scaled SpeechFrame front ends combined with vertically scaled application servers and database servers. Implementing load balancers in active/standby configuration guarantees 99.999% service availability.

Using a single chassis, up to 8 E1/T1 connections and up to 600 SIP channels can be delivered. Using compactPCI, SpeechFrame can control up to 480 TDM and 1.200 SIP channels.

Applications can be deployed on a centralized site on several application servers. Adding load balancers in front of the application servers provides load balancing and service availability control. Whenever there is a faulty application server, requests will not be sent to that machine anymore, leaving the total solution fully operational.

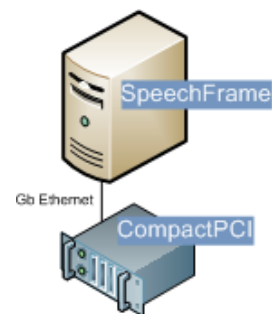
3.1. Boxed Solution

In a boxed solution all functionality is embedded into a single chassis. The solution covers all building blocks into one system, from SpeechFrame Physical Layer up to the Dialogue Applications and databases.

The maximum capacity for a boxed solution, using a single chassis, is 240 TDM and 600 SIP channels combined.

3.1.1. Increasing capacity

Other than installing PCI telephony boards into the chassis, an external compactPCI chassis can be used to increase capacity. The compactPCI cards have on-board CPU and run autonomically. The cards will boot and are then controlled over Ethernet by SpeechFrame.

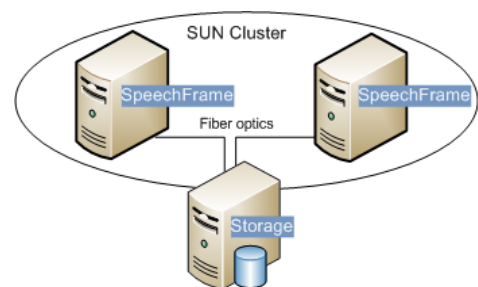


The maximum capacity using one compactPCI telephony board in combination with one SpeechFrame system is 480 TDM and 1200 SIP channels. The compactPCI chassis Comsys has chosen, has three redundant hot-swap power supply units (PSU) and can hold up to 8 telephony boards. One PSU can supply power to all 8 boards.

3.1.2. Adding redundancy

Using a single system can be vulnerable for complete service failure in case of outage of the system or any component. A second identical system can be added to achieve redundancy.

Placing the systems into a cluster guarantees high availability. Database and message storage will be placed on an additional storage unit which is connected with optical fiber.



Added redundancy using cluster and external storage, and the use of compactPCI telephony boards to increase capacity per unit can be combined in the solution.

3.2. Distributed Architecture

For large scale solutions and geographically separated POPs, the solution can be scaled vertically. The SpeechFrame core components will be installed on front end servers (POPs), the dialogues on a centralized site separated from the database servers. All back-end systems need to be reachable from the central site.

3.2.1. Front End Servers

The front end servers terminate TDM and/or SIP traffic on location. The locations can be anywhere and is not restricted to a city, country or continent for that matter. The single requirement is that the POPs can reach the centralized application servers. All service logic can be retrieved from the central site. Business logic is usually available in the back-end systems that are queried from the dialogues via the interface applications.

The POP has no logic regarding the dialogues whatsoever. In case of outage of a POP, or TDM/SIP connections towards the POP, traffic can be rerouted to any other POP. The latter in its turn retrieves the dialogues that normally run on the faulty POP and services should not be affected.

3.2.2. Application Servers

In order to achieve high availability, the application servers on the central site are installed N+1. Each application server can serve up to 1.500 dialogue instances under mass calling load circumstances. The servers are located behind two active/standby load balancers which route requests to the available application servers.

New or changed dialogues can be deployed during operational hours without any service disruption. Using version numbering in dialogue deployments, old and new versions can be deployed simultaneously. Changing the start-URL mapped to DNIS on the POP switches from old to new version dialogues on the fly.

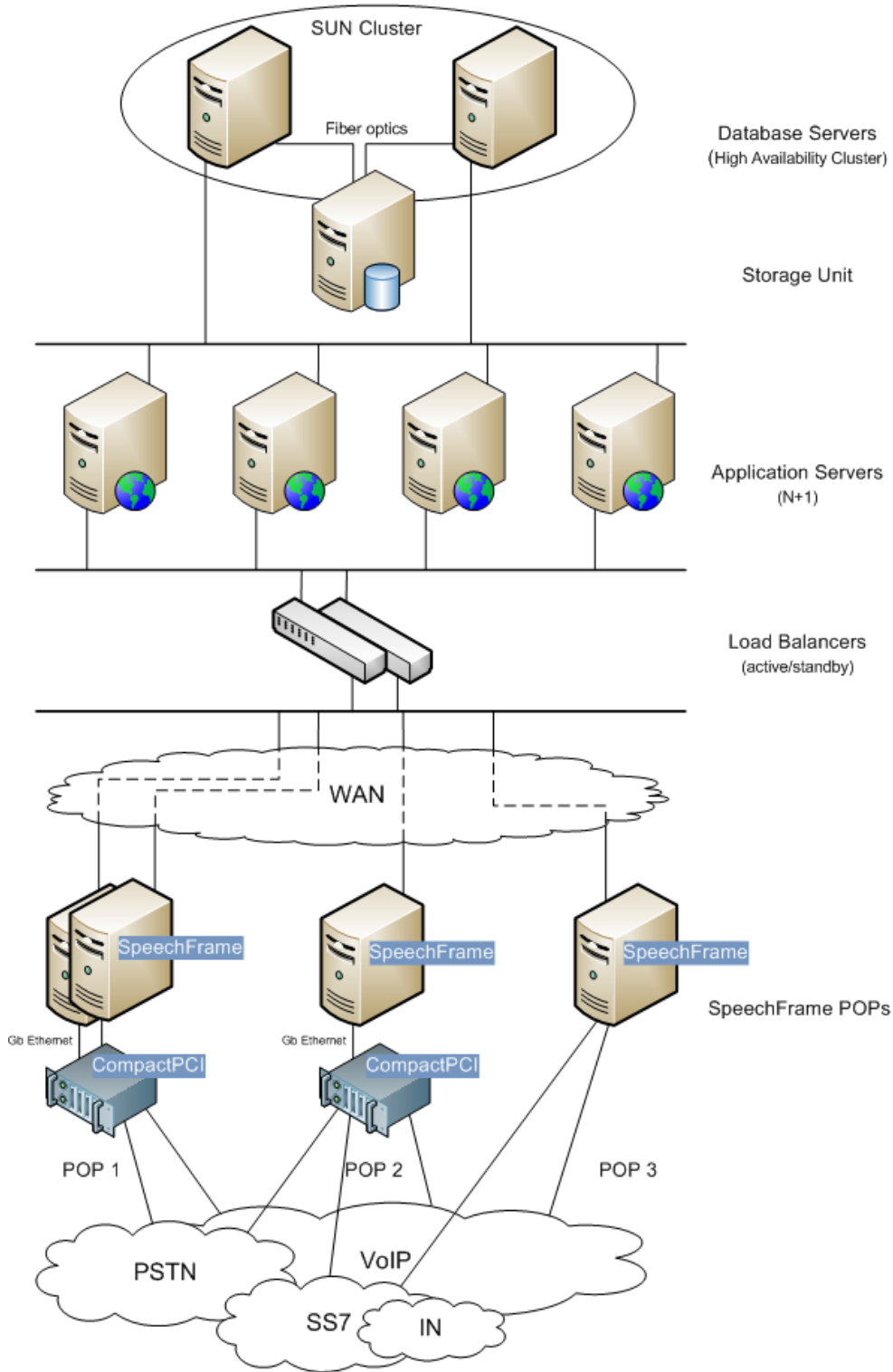
3.2.3. Database & Storage Servers

Centralized database servers are used to store statistical information. The CDRs and EDRs are inserted from the application servers. Next to statistical information, dialogue settings (i.e. call center routing numbers, open and closed hours, content references) are stored in the central database.

For dialogues that need high storage capacity (i.e. Ring Back Tone application), an Apache web server can serve as static content provider.

To achieve high availability, the central database & storage servers are placed in high availability cluster using an external storage unit.

3.2.4. Fully Distributed Example



Appendix A. Protocols and Approvals

Provided by Aculab. Comsys' preferred telecom board hardware partner.

SS7 protocol stack coverage

SS7 protocol	National and international variants	Specification compliance
MTP 2 (message transfer part layer 2)	ITU-T, ANSI, China	Q.703 (1996/white book); ANSI T1.111 (1996); China GF001-9001 (1990)
MTP 3 (message transfer part layer 3)	ITU-T, ANSI, China	Q.704 (1996/white book); ANSI T1.111 (1996); China GF001-9001 (1990)
ISUP (ISDN user part)	ITU-T, ANSI, ETSI, UK, China	ITU-T ISUP (1997/white book); ANSI ISUP T1.113 (1995); Q.767 International ISUP; China ISUP YDN-038 (1997); ETSI ISUP V4 (2001); UK ISUP (2001); user definable variants ¹
SCCP (signalling connection control part)	ITU-T, ANSI, China	Q.711-Q.714 (1996/white book); ANSI SCCP T1.112 (1996); China SCCP GF010-95
TCAP (transaction capabilities application part)	ITU-T, ANSI, China	Q.771-Q.774 (1996/white book); ANSI TCAP T1.114 (1996); China TCAP GF011-95

¹ Aculab's SS7 software provides a flexible option through which the user can define other national and international ISUP variants to meet specific needs.

IP protocols and signaling stacks

Protocol	IETF Specification	Feature description
SIP (session initiation protocol)	RFC 3261	Session initiation protocol SIP on UDP and TCP SIPS (SIP over TLS)
	RFC 3262	Reliable provisional responses
	RFC 3310	SIP authentication
	RFC 2327	Session description protocol (SDP)
	RFC 3665	Basic call flow examples
	RFC 3666	SIP/PSTN call flows
	RFC 3264	Offer/answer model with SDP
	RFC 3725	Third party call control best practices
	RFC 3515	The REFER method
	RFC 3204	MIME media types for Q.SIG/ISUP
	RFC 2976	INFO method
	RFC 3891	Replaces header
	Draft-ietf-sipping-service-examples-09	Hold and transfer
	Draft-ietf-sipping-cc-transfer-06.txt	Blind transfer for SIP
	RFC 3892	Referred by header
	RFC 3261	TCP support
	RFC 3581	Symmetric signalling ports
	Draft-ietf-mmusic-sdescriptions-12	Secure RTP support
	RFC 4028	SIP session timers
	RFC 3265	Subscribe/specific event notification
RFC 3311	UPDATE method	
RFC 3489	STUN API	
MRCP (media resource control protocol)		MRCP v1, draft 7
		MRCP v2, draft 11
Protocol	ITU-T Specification	Feature description
H.324M	H.324M	Including support for H.223
3G-324M		Including support for H.223

Approvals for connection to the public network

Country	Protocol	Approval standard	Approval number	Additional notes
Australia	Q.931/Q.932	AS/ACIF S038	Self declaration	E1
Australia	Q.931/Q.932	AS/ACIF S038	Self declaration	E1
Brazil	Brazilian ISDN	Anatel	0030-06-1140	E1
Brazil	Brazilian ISDN	Anatel	0032-06-1140	E1
Brazil	Brazilian ISDN	Anatel	0028-06-1140	E1
Brazil	Brazilian ISDN	Anatel	0029-06-1140	E1
Brazil	Brazilian ISDN	Anatel	0027-06-1140	E1
Canada	Aculab T1 protocols	CS03 part 8	2789A-AC5200	T1
Canada	Aculab T1 protocols	CS03 part 8	2789A-PCIEX	T1 – fitted with DSP for CAS/SS7 if applicable
Canada	Aculab T1 protocols	CS03 part 8	2789A 12217	T1 - fitted with DSP if applicable
Canada	Aculab T1 protocols	CS03 part 8	2789A-AC7070	T1
China	Q.931/Q.932	Chinese ISDN	12-7170-060345	E1
China	Q.931/Q.932	Chinese ISDN	15-5288-020439	E1
China	Q.931/Q.932	Chinese ISDN	15-5288-020440	E1
China	Q.931/Q.932	Chinese ISDN	12 7170 050931	E1
China	Q.931/Q.932	Chinese ISDN	2005021607000001	E1
China	Q.931/Q.932	Chinese ISDN	15-5288-020441	E1
EU wide	Q.931/Q.932	TBR4	Self declaration under RTTE	E1 protocol also referred to as Euro or ETSI ISDN
Hong Kong	ITU T1	HKTA2015	IN606049	T1
Hong Kong	ITU T1	HKTA2015	IN606048	T1
Hong Kong	ITU T1	HKTA2015	IN406047	T1
Hong Kong	ITU T1	HKTA2015	IN403011	T1
Hong Kong	ITU T1	HKTA2015	IN603014	T1

Hong Kong	ITU T1	HKTA2015	IN603018	T1
Hong Kong	ITU T1	HKTA2015	IN603017	T1
Hong Kong	ITU T1	HKTA2015	IN405010	T1
India	Q.931/Q.932	TEC	TEC/WR/I/CTI-01/02/052.SEP 04	E1
Japan	INS 1500	Japan	07225004/AA/00	T1
Japan	INS 1500	Japan	07225005/AA/00	T1
Japan	INS 1500	Japan	07225006/AA/00	T1
Japan	INS 1500	Japan	07225002/AA/00	T1
Korea	Q.931/Q.932	Korean requirements	TE-C99/K900-03-00093	E1
Korea	Q.931/Q.932	Korean requirements	TE-C99/K900-03-00091	E1
Korea	Q.931/Q.932	Korean requirements	TE-C99/K900-03-0090	E1
Korea	Q.931/Q.932	Korean requirements	TE-C11/K031-05-003	E1
Korea	Q.931/Q.932	Korean requirements	T-C99-00-1220	E1
Malaysia	Q.931/Q.932	TPS-013-01	CETS/394B/0506/T	E1
Malaysia	Q.931/Q.932	TPS-013-01	ISDA/48A/0603/S	E1
Malaysia	Q.931/Q.932	TPS-013-01	ISDA/47A/0603/S	E1
Malaysia	Q.931/Q.932	TPS-013-01	ISDA/51A/0603/S	E1
Malaysia	Q.931/Q.932	TPS-013-01	Awaiting approval number	E1
Malaysia	Q.931/Q.932	TPS-013-01	ISDA/46A/0603/S	E1
Malaysia	Q.931/Q.932	TPS-013-01	CETS/393B/0506/T	E1
Malaysia	Q.931/Q.932	TPS-013-01	ISDA/50A/0603/S	E1
Mexico	Q.931/Q.932	Cofotel	RCPACAC04-712	E1
Mexico	Q.931/Q.932	Cofotel	RCPACAC04-651	E1
New Zealand	Q.931/Q.932	PTC 232	PTC232/06/001	E1
New Zealand	Q.931/Q.932	PTC 232	PTC220/02/029	E1
New Zealand	Q.931/Q.932	PTC 232	PTC220/02/032	E1
New Zealand	Q.931/Q.932	PTC 232	PTC220/02/030	E1
New Zealand	Q.931/Q.932	PTC 232	PTC 217/05/007	E1
New Zealand	Q.931/Q.932	PTC 232	PTC220/02/031	E1

Singapore	Q.931/Q.932	IDA TS ISDN-PRA	G0373-06	E1
Singapore	Q.931/Q.932	iDAS ISDN2	ISDN2-0631-2003	E1
Singapore	Q.931/Q.932	iDAS ISDN2	ISDN2-0633-2003	E1
Singapore	Q.931/Q.932	iDAS ISDN2	ISDN2-0630-2003	E1
Singapore	Q.931/Q.932	iDAS ISDN2	ISDN2-0632-2003	E1
Singapore	Q.931/Q.932	iDAS ISDN2	ISDN2-T0314-2005	E1
South Africa	Q.931/Q.932	TBR4	SS-743.01	E1
South Africa	Q.931/Q.932	TBR4	SS-425.01	E1
South Africa	Q.931/Q.932	TBR4	SS-427.01	E1
South Africa	Q.931/Q.932	TBR4	SS-424.01	E1
South Africa	Q.931/Q.932	TBR4	TE-2004/189	E1
South Africa	Q.931/Q.932	TBR4	SS-423.01	E1
USA	Aculab T1 protocols	FCC part 68	5TCXDNANPMXPCIX	T1 - fitted with DSP if applicable
USA	Aculab T1 protocols	FCC part 68	5TCXDNANPM4MODT1	T1 - fitted with DSP if applicable
USA	Aculab T1 protocols	FCC part 68	5TCXDNAPMXT1MOD	T1

Worldwide protocol coverage

Country	Protocol	Protocol type	Additional notes	Aculab protocol stack
Argentina	R2 CAS	CAS	.	R2T1
Australia	TS038	CCS	.	ETS 300
Australia	P2	CAS	TS003/TPH1271/R2D	R2T1
Belgium	National R2	CAS	.	R2T1
Belgium	National R2 DTMF	CAS	.	BELGU
Brazil	Euro ISDN	CCS	.	ETS 300
Brazil	MFC R2	CAS	Brazil 5C	R2T1
Canada	T1 Robbed bit	CAS	.	T1RB
Chile	MFC R2	CAS	.	R2T1
China	R2	CAS	China#1	R2T1
China	Chinese ISDN	CCS		ETS300
Colombia	R2	CAS	.	R2T1
Croatia	R2	CAS	.	R2T1
Czech Republic	R2	CAS	.	R2T1
Czech Republic	MFC R2	CAS	Type K	R2T1
Denmark	National MFC R2	CAS	.	R2DK
Egypt	MFC R2	CAS	.	R2T1
EU wide	Euro ISDN	CCS	.	ETS 300
Finland	R2	CAS	.	R2T1
France	MF R1 Socotel	CAS	.	FMFS
France	VN6	CCS	.	ETS 300
Greece	OTE 4	CAS	4-bit CAS	OTE4
Greece	OTE 2	CAS	2-bit CAS	OTE2
Hong Kong	HKTA 2015	CCS	.	ETS 300
Hong Kong	HKT 2018 Robbed bit	CAS	T1HK; AMI or B8ZS encoding	T1HK
India	MFC E&M	CAS	.	R2T1
India	MFC R2	CAS	Type 1/2/3	R2T1
Indonesia	R2 (Q.421)	CAS	Ericsson loop signalling	R2T1
Indonesia	SMFC R2	CAS	Semi-compelled	IEM

Iran	R2	CAS	3-bit decadic	R2T1
Israel	ETS 300	CCS	.	ETS 300
Israel	MFC R2	CAS	Israel R2	R2T1
Italy	I701	CAS	.	I701
Japan	INS 1500	CCS	.	INS1500
Jordan	R2	CAS	.	R2T1
Korea	Euro ISDN	CCS	.	ETS 300
Korea	R2	CAS	.	R2T1
Kuwait	R2	CAS	.	R2T1
Latvia	MFC R2	CAS	.	R2T1
Malaysia	MFC R2	CAS	.	R2T1
Malaysia	MFC R2	CAS	.	IEM
Malta	MFC R2	CAS	.	R2T1
Mexico	R2	CAS	.	R2T1
Netherlands	ALS70D	CAS	T11-53E	ALSN/ALSU
Netherlands	MFC R2	CAS	.	R2T1
New Zealand	TNA134	CCS	Q.931	ETS 300
Norway	National MFC R2	CAS	.	R2T1
Peru	MFC R2	CAS	.	R2T1
Philippines	R2	CAS	.	R2T1
Poland	EuroISDN	CCS	.	ETS 300
Poland	MFC R2	CAS	.	R2T1
Portugal	MFC R2	CAS	.	R2T1
Sierra Leone	MFC R2	CAS	.	R2T1
Singapore	IDA TS ISDN2	CCS	.	ETS 300
Singapore	Fetex	CCS	.	FETEX
Singapore	MFC R2	CAS	.	R2T1
Singapore	MFC R2	CAS	.	IEM
South Africa	Euro ISDN	CCS	.	ETS 300
South Africa	MFC R2	CAS	.	R2T1
Spain	MF R1 Socotel	CAS	.	SMFS
Sweden	CAS extension EL7	CAS	Ericsson ASB/voicemail	EL7

Sweden	P8	CAS	P8 DDI and P7 non-DDI option	P8
Taiwan	MF R1	CAS	Modified	T1RB
Thailand	National R2 DTMF	CAS	.	R2T1
Turkey	R1	CAS	.	E1LS
UK	DASS2	CCS	.	DASS
UK	DPNSS	CCS	.	DPNSS
UK	BT/MCL Interconnect	CAS	Asymmetrical	BTMC
UK	BT Callstream	CAS	SIN 205/356	BTCU/BTCN
UK	PD1	CAS	MCL PD1/DC5A	PD1
USA	AT&T	CCS	TR41459	ATT-T1
USA	DMS 100	CCS	Nortel DMS (T1)	DMS100
USA	National ISDN 2	CCS	NI1 and NI2	NI2
USA	National ISDN 2	CCS	NFAS (with D-channel back-up)	NI2
USA	T1 robbed bit	CAS	.	T1RB
Worldwide (ex USA)	E1 line side CAS	CAS	AT&T Definity and Nortel Meridian	E1LS
Worldwide	MFC R2	CAS	Q.421/Q.441	R2T1
Worldwide	SS5	CAS	CCITT SS5 (C5)	SS5
Worldwide	Decadic CAS	CAS	Generic use with PBXs	R2T1
Worldwide	E&M type A	CAS	Ericsson DC5 and E&M options	EEMA
Worldwide	30DLI	CAS	NEC PA-30DTS	30DLI
Worldwide	SS7	CCS	ITU-T: ISUP Q.767; TCAP Q.771-Q.774; SCCP Q.711- Q.714; MTP Q.703, Q.704, Q.707	ISUP/TCAP
Worldwide	Q.SIG	CCS	.	QSIG