SPEAKERBUS

i series

“Endpoint Range”

www.speakerbus.com

i series

Speakerbus iSeries desktop devices enhance collaboration through instant voice communications enabling people to work effectively as teams across functional and geographical boundaries to facilitate rapid decision making, through efficient and effective communications.

This brochure helps you determine which Speakerbus desktop devices, are right for your organisation, where best to use these endpoints, and how they can help you maximize your overall investment in Speakerbus iSeries range of collaboration solutions.

Contents

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iSeries Endpoint Overview

Our desktop devices are designed to leverage the evolution of unified communications and service oriented architectures by providing:

- Advanced productivity features - multiple call handling with automatic stacking/unstacking of prioritised calls.
- A feature set that delivers traditional functionality; whilst creating new functions and sets high standards for one touch operation, ease of use and operational flexibility.

Strengthen collaboration throughout teams and communities helps businesses:

- React to market and environmental events before their competitors.
- Improve the flow of communication through instant collaboration with local, regional and global counterparties providing a platform for more informed decision making.
- Distribute information to multiple parties simultaneously, which means an increasingly informed user community, that leads to enhanced collaboration and drives business and workflow efficiencies.

Speakerbus has been developing mission critical communication and collaboration solutions for the most demanding markets for the last 28 years. Speakerbus solutions are installed in every major financial centre throughout 30 countries.

In today's economy, your business must meet the needs of a wide range of users with different communications styles and individual requirements. Speakerbus provide a range of endpoints from the sophisticated iTurret through to entry level single channel intercom, the iD712.

The core services offered by our endpoints:

- Intercom Functionality
  - Point 2 Point
  - Group calls
  - Answerback calls
  - Hoots
  - Intercom Privacy
  - Last Intercom Number Redial
  - Intercom Speed dials Personal & Group
  - Mute
  - Talk Latch
- Line Sharing
  - Common Lamping
  - Barge-In
  - Privacy
- Remote Working
  - SIP call routing
  - Line networking for Private Circuits
- Voice Nets
  - Hoot
  - Intercom
  - PLAR/Crash Net

iSeries Intercom Endpoint Range.

iManager CS Endpoints

Speakerbus' endpoints follow a simple methodology of user-centred design; exploiting visual elements; contextual soft keys and intuitive menus, supporting agile responses in every situation. The user interface and the embedded menu are designed to ensure the user is able to navigate efficiently to the required functionality, using the minimal number of key presses. An intuitive set of context sensitive soft keys guides the user to the specific functionality they require, whether they are active on internal communications or are customising the device to personalise their experience.

Single Channel Intercom - iD712

The iD712 provides core functionality at an entry level cost to the range of iSeries intercom endpoints. It can be used to link user groups and support teams with breaking news and group conferencing. Together these tools lead to an increasingly informed user community, which in turn enhances collaboration driving business and workflow efficiencies.

The user interface utilises a graphical display and takes full advantage of the colour screen, exploiting visual elements; contextual soft keys and intuitive menu's, supporting agile responses in every situation.

Handset support allows dual calling on the station to increase the user's capabilities while the advanced user interface will automate many features such as call prioritisation and the stacking/un-stacking of calls.

The iD712 colour display in tandem with under the hood ability to automated prioritisation of interactions, ensures the user's attention is quickly directed to the highest priority internal communication. Whether distributing information to multiple parties or evaluating informational feeds the iD712 makes highly effective use of its colour display. User's attention is quickly directed to events or changes in state.

Special attention is given to the use of display colours, graphics, fonts and text within the displays. Graphical icons are used to convey to the user the internal call types they are active upon, using the iSelector function, they can easily switch between active calls as and when market and environmental events dictate.

From left to right Point to Point, Group Call and Voice Services call type graphical icons.

A stylish design belies a module enclosure built for reliable performance in the most rigorous environments.

- High Resolution Colour Screen with Stylised graphics, fonts and icons to guide the user.
- Full Dial Pad with Soft Function Keys and Direct Function Keys.
- Multiple Audio Interfaces: Embedded Microphone with Optional Gooseneck Microphone, Handset and Headset Support.
- Multiple Call Handling with Automated Stacking and Un-Stacking of Prioritised Calls.
- LED Indicators for Call State Awareness and Optimal Microphone Usage.
- Access to Multiple Directories, Including Personal Directory Customisation.
- 32 User Defined Speed Dials
- Standard Ethernet Connection with PoE (Class II) Support.

Intercom Functionality

- Point 2 Point
- Group Calls
- Answerback Calls
- Hoots
- Intercom Privacy
- Last Intercom Number Redial
- Intercom Speed dials Personal & Group
- Mute
- Talk Latch

Handset - Available handsets incorporate a software controlled mute or push to talk operation. A handset volume control is included for earpiece volume adjustment along with a noise cancelling microphone.

Headset - The iD712 support for headset operation, enabling users to multi-task whilst managing internal communications with the added benefit of their voice being consistent. For compatible headsets please refer to the accessories section.

Station Power - The iD712 can be powered using a suitable Power Over Ethernet infrastructure that conforms to 802.3af Local power can also be supported using a suitable mid span supply and power splitter, see accessories section for further details.

- PoE Class II IEEE 802.3af
- Power rating typically: 3W; Maximum: 6.49W

Sleek tabletop design with open mic or gooseneck microphone.

Alpha numeric keypad & fixed function keys.

Icon based indicators & high resolution colour screen.

Eight Channel Trader Endpoint - ZERO8 (SE 708)

The ZERO8 is a versatile multi-channel speaker device for trading applications. The ZERO8's collaborative interactions enable traders to communicate with their advisors, counterparties and clients using Private Wire, Intercom, Hoot 'n' Holler, Telephony and Broadcasts. The iManager Portal enables plug 'n' trade provisioning for faster deployments. The portal continually monitors the network rapidly identifying new devices and synchronises known endpoints with 'live' updates. The ZERO8 achieves high level speaker output while retaining an compact form factor using a tuned speaker enclosure and dual drivers.

Interconnected Parties

Featuring eight, user-assignable speaker channels that support both automatic and manual private wires; the ZERO8 enables users to simultaneously work with multiple counterparties or clients. ZERO8 supports speaker/gooseneck microphone operation and speaker/handset modes allowing the user to tailor their activities to the type of trading. The user interface supports rapid assignment and clearing of line to speaker and lists all available lines in the directory for ease of access.

As part of the iSeries product family, the ZERO8 shares private lines with other speaker and dealerboard endpoints from Speakerbus with barge in, supervisory lamping, privacy and global muting functions supported.

Support colleague's phone lines with up to 200 prioritised bridged iCS appearances for total call coverage. Dial local and external parties directly from the device without switching over to a dealerboard or phone.

Multi-site trading operations benefit from the line networking provided by iSeries gateways to extend coverage across multiple locations and time-zones. An effective toll bypass solution using the enterprise infrastructure can deliver considerable cost savings.

Enhanced Internal Collaboration

Providing the trader with a choice of collaborative interactions the ZERO8 includes telephony[1], hoot, intercom, and broadcasts, supporting complex trades whilst aiding the faster closure of deals. A key feature of the ZERO8 is its ability to offer 8 channels of always on audio ensuring access and delivery of morning briefings, TV, audio broadcasts and breaking news along with collaborative group calls.

With multiple speed dial pages and search enabled directories the ZERO8 enables fast and extensive access to colleagues with the instant connectivity of global intercom and group based collaboration.

Designed for the Trader

The ZERO8 user interface is designed for the demanding needs of the trading environment, utilising a clean, simplified user interface. 'In key' channel status lamps are augmented by iconography to convey a clear status of communication activities across all speaker channels. Splash screens provide calling party information for intercom activities while status icons and activity prompts, guide the user through all operations. Physical function keys for call clear, hold and access to the home screen mark out

1 Access to telephony features requires Cisco Unified Communications manager and a iManager Communication Server

the ZERO8 as a thoroughbred trading endpoint.

Compliance Ready

Centralised management ensures adherence to company policy and governance via user permissions, partitioning and auditing. Permissions and policies may be assigned to individual or user groups for the ZERO8 and cover access to all call types and lines.

Speakerbus' next generation iSeries endpoints are compliance-ready devices with voice and call data recording interfaces. The iSeries products support an active recording interface that transmits Call Detail Records (CDR) and associated media that relates to events happening on an individual device.

The active recording method ensures that all calls, in-call events and specialised voice services are captured by the recorder. The detail available in the CDR ensures that search and replay of recordings is optimised and that a detailed autopsy of call progress can be made as calls transit between user functions such as conference, switching to headset, handset or speaker or initiating talkback on a speaker channel.

The active recording interface is dual redundant and configured by centralised policies that are distributed and enforced by the iManager Portal and centralised management system.

Key Features

- 8 User Assignable Speaker Channels
- High Resolution Colour Screen
- Telephony - Make Calls, Answer Bridged Call/Line Appearences (up to 200), Place Calls on Hold and Redial features
- Call Registers - All Calls, Missed Calls, Received Calls and Placed Calls.
- Full Dial Pad
- Multiple Audio Interfaces: Optional Gooseneck Microphone, Handset and Headset Support.
- LED Indicators for Call State Awareness and Optimal Microphone Usage
- User configurable via embedded menu system

Private Wire Functionality

- Support for Eight Private Wires - ARD and MRD
- Choice of Speaker Channel Muting
- Private Wire Privacy
- Latching -Push To Latch and Tap Latch
- Easily switch between Handset and Gooseneck

Handset & Headset

The ZERO8 supports optional handset or headset operation. Please refer to the accessories section for compatibility information.

Endpoint Power

The ZERO8 can be powered using a suitable Power Over Ethernet infrastructure that conforms to 802.3af. Local power can also be supported using a suitable mid span supply.
See accessories section for further details.

- PoE Class III IEEE 802.3af
- Power rating typically: 6.49W; Maximum: 15.4W

Rotary volume channel control, navigation and mode keys.

Illuminating LED indicators, home key & high resolution colour screen.

Alpha numeric keypad & fixed function keys.

Dealerboard Intercom - iTurret

Powered by iManager the iTurret dealerboard provides a flexible approach to trader voice in a modular design.

The most advanced station in the iSeries range, iTurret delivers instant point to point, group and multipoint intercom calls, delivering enhanced and more informed group communications linking traders, user groups and support teams with breaking news and group conferencing. Using this tool set leads to an increasingly informed user community, which in turn enhances collaboration driving business and workflow efficiencies.

- Delivers High Performance Multi-Channel Voice.
- 8 Channels of Speaker (Hoot 'n' Holler).
- Full Dealerboard Functionality.
- Multiple High Resolution Colour Screen.
- Full Dial Pad with Soft Function Keys and Direct. Function Keys.
- Multiple Audio Interfaces: Embedded Microphone with Optional Gooseneck Microphone, Twin Handset and Headset Support.
- LED Indicators for Call State Awareness and Optimal Microphone Usage.

Intercom Functionality

- Point 2 Point
- Group Calls
- Answerback Calls
- Hoots
- Intercom Privacy
- Last Intercom Number Redial
- Intercom Speed dials Personal & Group
- Mute
- Talk Latch

Expansion Modules

The main module can be augmented with expansion units (side cars) for additional soft keys and speaker (Hoot 'n' Holler) channels if required.

iTurret 8 channel speaker expansion module – iE801

The iE801 offers integral speaker paging, facilitating up to 96* speaker channel assignments, spread across four pages of speaker channels. Multiple pre-set pages of speaker channels can be defined, with rapid change to speaker programming and automatic reconnection of speaker channels.

iTurret 16 key expansion module – iE816

The iE816 button expansion module provides extra buttons, soft keys and twin additional colour displays.

iTurret user interface

The iTurret user interface has been designed to convey a high degree of inherent familiarity to the user. Many of the best features utilised by leading cell phones have strongly influenced the iTurret user interface, in particular the operational aspects of the integral ergonomic navigation keys.

For a comprehensive overview of the iTurret and its companion expansion modules please see the iTurret detailed product brochure.

Dealerboard module - iTurret

Innovative virtual trader voice endpoint - ARIA

Virtual and physical trader voice – Get the best of both worlds.

Key Benefit

- Reduced cost of ownership
- Greater workforce mobility
- Enhanced business continuity
- Compliance & security

With over 30 years of proven track record of delivering innovative voice collaboration solutions to financial institutions world-wide, we deliver the best of both worlds - virtual and physical trader voice endpoints.

In response to the rising demand for cloud based technologies, we present a solution that enables greater workforce mobility and enhanced disaster recovery, with reduced total cost of ownership.

iManager ARIA is our new virtual endpoint, providing the same global hoot, advanced telephony and private line capabilities enjoyed by iTurret dealerboard clients worldwide today.

Trader voice capabilities are accessible from the browser, removing the need to install any software.

Choose who you trade with in a secure, compliant and cost-effective manner.
Powered by HTML5 technology, the ARIA user experience adapts to desktop and mobile device environments, providing unrivaled versatility.

Let us help you transform your trading communications landscape with ARIA.

Business Benefits

- Reduced total cost of ownership
- Remove the need for proprietary hardware from business continuity locations.
- Remove day two costs with iManager Portal for moves, adds and changes.
- Leverage Bring Your Own Device (BYOD) and remote desktop services.
- Benefit from a single enterprise platform to streamline operations and reduce support overheads.
- Transparent support for existing voice recording and analytics solutions.
- Distribute digital and SIP private lines across the organization between physical devices and ARIA users.

Web interface overview - ARIA

Web interface - SLATE^{ux} interactions are clear and ease to use

Greater workforce mobility

- Instantly access trader voice functionality via a computer or mobile device, using a web browser.

Enhanced business continuity

- Access trader voice functionality from PCs and BYOD's via a internet connection.
- Instant access 'free seating' using common user credentials across Speakerbus' platform.

Compliance

- Provide continuous compliance coverage though a common API across Speakerbus' physical and virtual solutions.
- iTurret dealerboard users inherit all permission and recording policies.

Security

Secure, encrypted user sessions prevent interception or snooping.

High Scalability and Resilience

- Enterprise grade CLOUDBASE technology that is engineered to scale to 10,000 users.
- A mature SIP technology stack deployed across Speakerbus' iSeries since 2007 with proven interoperability with leading UC vendors.
- Multi-server distributed processing that is built to deploy in resilient data-center infrastructures.
- Secure containerised applications for each discrete user session with distributed media processing.
- High availability application, database and gateway services.
- Maximise existing Unified Communications (UC) investment
- Certified interoperability with Cisco, Avaya & Mitel.
- Compatible SIP that is proven on Cisco for 4 generations of CUCM.
- Certified as compatible with Avaya since 2010.
- Certified as compatible with Mitel since 2012.
- Proven, advanced centralised management
- Speakerbus' iCentralised Management System (iCMS) is currently in service at all iSeries client sites world-wide, including tier1 banks, brokerages and asset management firms.
- Tens of thousands of endpoints and user profiles are managed by this platform.

Infrastructure

User Authentication: The Speakerbus endpoint devices natively supports two levels of access authentication in the form of user and administrator levels. In addition to normal user access, an administrative level is provided in order to restrict access to specific embedded menus on the Speakerbus endpoint devices intended for configuration and support tasks. Both access levels are password controlled and are maintained using the iManager web portal management application.

Encryption: Password data is safeguarded using cryptographic hash functions and encryption techniques.

Quality of Service Techniques (QoS)

The Speakerbus endpoint devices are adaptive in nature, they will add delay when packets get delayed or dropped, seeking the best possible performance under prevailing network conditions. Most switching and routing devices currently in use will be 'QoS' capable, however the network design as a whole and the detailed configuration of the switches and routers work together to determine if the network will provide the service levels to make VoIP practical.

QoS techniques are employed to compliment the network design and these are supported by Speakerbus endpoint devices:

- Bandwidth optimisation techniques: VAD (Voice Activity Detection).
- Diffserv (RFC 2474) – Type of service field configurable.

Trading Compliance

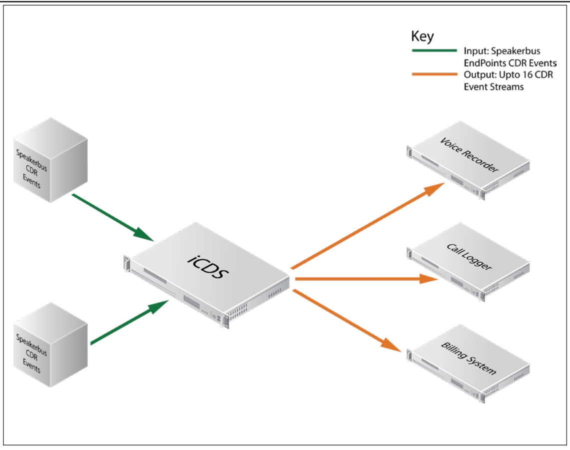
Compliance is implemented via "active recording," on the Speakerbus endpoint devices whereby its own premixed recording streams are sent direct to the recorder along with the associated call information.

Compliance Reporting: The Speakerbus endpoint device call information is collated by the Speakerbus iCDS (iManager Call Data Service). This is a Microsoft® application that unifies all Speakerbus endpoint device call information. The iCDS application runs as a service to concentrate up to 16 CDR call information streams that can be shared out to multiple third-party applications via a standard IP socket interface. The Speakerbus endpoint device call information stream is a standardised protocol and data schema that can be utilised by applications offered through certified specialist Speakerbus vendors.*

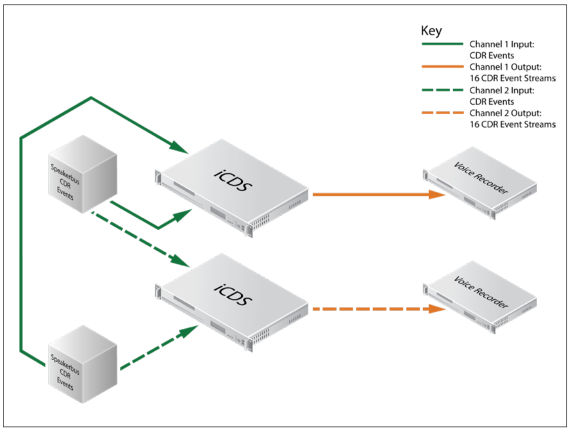
Compliance Recording: The recording of Hoot 'n' Holler, Intercom, Private Wires and Telephony traffic is facilitated with a dedicated RTP stream or through a Speakerbus digital or an analogue gateway to an approved voice recorder. Dual iCDS can be deployed for compliance resilience requirements. Speakerbus endpoint device call traffic can also be monitored with a voice activated recording (VOX) feature of the recorder to minimise storage requirements.

Compliance Resilience: Speakerbus endpoint devices have the ability to support either a 1+1 (active/active) or N+1 (active/passive) compliance mechanisms. Up to 6 dedicated recording outputs, in RTP form, can be defined to approved IP enabled recorders depending on compliance requirements. If speaker expansion modules (iE801's) are deployed an additional 4 dedicated RTP recording streams are available, per module, for compliance. RTP recording streams can be minimised by mixing handset and speaker channels directly at the Speakerbus endpoint device as a means to minimise traffic and recorder capacity requirements.

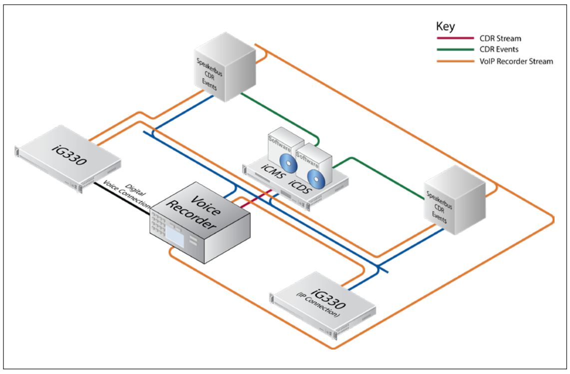
Each Speakerbus endpoint device provides a stream of call and voice activity events to the iCDS application while providing a copy of the voice stream to the recorder. Voice streams can be minimised by mixing handset and speaker channels directly at the Speakerbus endpoint device as a means to minimise traffic and recorder capacity requirements.



iCDS concentrator API architecture



iCDS for replication of real time API's to multiple destinations



iCDS distribution of API's and associated Audio events to Voice Recorders (either IP or Digital based)

Accessories

Handset

Speakerbus Handset[1] – Momentary latching/latching Handset

Designed and developed directly for critical communications.

- Robust.
- Momentary latching bar and latching switch
- Volume Control.
- Noise cancelling system.

SE HSETM-C – Walker style momentary latching Handset
SE HSETL-C - Walker style latching Handset

Proven track record in Finance market.

- Robust.
- Momentary latching key.
- Noise suppression system.

Microphone

Close Talking Gooseneck Microphone

- Flexible talking position.
- Close talking microphone - removes external noise from surrounding environment i.e. busy trading floor or office.

iD712 Intercom Specifications

Channel Capacity
- 1 Channel shared using iSelector

Call Types
- Point 2 Point (P2P) Intercom
- Group Calls
- Answerback Calls
- Dynamic Audio Conference
- Hoot ‘n’ Holler-Broadcast

Intercom Functionality
- 32 Intercom Speed Dials
- Intercom Privacy
- Last Number Redial
- Directories - Global, Personal, Location and Group
- Intercom Call Registers - Placed, Received and Missed

User Features
- Access to 32 user defined Speed Dials
- Personal directory with up to 500 entries
- Corporate directory with up to 15000 entries
- Intercom group directory up to 2500 entries
- 4 Context sensitive soft keys

System Management
- Upgradeable operating firmware
- Managed using iManager iCMS
- SNMP status monitoring

Recording
- 1 Configurable IP recording stream
- 1 Resiliency IP recording stream
- Call Data Records (CDR) output

Network
- IP Addressing: dynamic or static
- Supported network protocols: Ethernet, IPv4, DHCP, TCP/IP, DNS, 802.1p, 802.1q and SNMP

VoIP Media
- SIP (using standards based RFC’s)
- RTP supported codecs: G.711 PCM (3.4KHZ) A-law/U-law, G.722 (48K), G.729 Annex A – CS-ACELP
- Voice on LAN: SbRTP[1]/DMVS[2]: 7KHZ enhanced voice bandwidth or 3KHZ low bandwidth
- Typical latency over LAN 6ms (using 1ms packet sizes)
- Voice on WAN: Unicast network supporting UDP
- Max Packet Loss on the LAN 5%
- Bandwidth optimisation techniques: VAD (Voice Activity Detection)
- Diffserv (RFC 2474) –Type of service field configurable

Input Devices
Standard
- Built-in open microphone
- Built-in loudspeaker

Optional
- Speakerbus pluggable gooseneck microphone
  - Close Talking Noise Cancelling Micro-phone (52-09-021) 550mm, 8.5mm (21.6 x 0.33 in) Diameter
- Speakerbus noise cancelling handset (SE HSETM-D)
- Third party handset/Headset Support

Voice Characteristics
- Loudspeaker Output:
  - 82dB(A) SPL @ 1KHz : 1M

Housing
- ABS plastic
- Detachable Steel Base Plate

Dimensions
- Width: 83mm
- Height: 105mm
- Depth: 220mm
- Weight: 540g

Power Requirements
- PoE Class II IEEE 802.3af
- Typical: 3W
- Maximum: 6.49W

Cable Requirements
- Ethernet Interface
  - Cable: Minimum Category 5e UTP Maximum length 100m
  - Connector: RJ-45

Interfaces
- Network interface 10/100BASE-T Ethernet auto sensing with PoE (RJ45 Socket)
- Handset interface port (RJ11 Socket)

Sleek tabletop design with open mic or gooseneck microphone.

Alpha numeric keypad & fixed function keys.

Icon based indicators & high resolution color screen.

Environmental
- Operating temperature 0°C – 35°C (32 - 95°F)
- Relative humidity 10% - 95% RH, non condensing
- Storage environment Temperature 0°C 60°C (32°F - 140°F), Humidity 10% – % RH

Declaration of Conformity
- EN60950-1, EN55022 and EN55024

1 Speakerbus Real Time Protocol
2 Dynamic Multicast Voice Services

SE 708 Eight Channel Trader Endpoint Specifications

Speaker Channels
- 8 Dynamic Channels
- Rotary Volume Controls
- Multi-Colour LED status indication 'in key'
- Tap Latch and PTT modes supported
- Auto Answer for Incoming Point to Point
- Answer to handset supported
- User Programmable

Call Types
- Hoot 'n' Holler
- Group Calls
- Answerback Calls
- Audio Distribution
- Point to Point Intercom
- Private Line Automatic Ring Down (PLAR)
- Private Line Manual Ring Down (MRD)
- Telephony[1]

Intercom Functionality
- 32 Intercom Speed Dials
- Intercom Privacy
- Last Number Redial
- Directories - Global, Personal, Location and Group
- Intercom Call Registers - Placed, Received and Missed

Telephony Functionality[1]
- Last Number Redial
- Telephony Call Registers - Placed, Received and Missed
- Hold[2]
- Up to 200 Answer Bridged Call/Line Appearances

Appearances
- Up to 439 Appearances (DMVS, iCS intercom and iCS/Cisco telephony)

Recording
- 3 Configurable IP recording streams
- 3 Resiliency IP recording streams
- Call Data Records (CDR) output

Network
- IP Addressing: dynamic or static
- Supported network protocols: Ethernet, IPv4, DHCP, TCP/IP, DNS, 802.1p, 802.1q and SNMP

VoIP Media
- SIP (using standards based RFC's)
- RTP supported codecs: G.711 PCM (3.4KHZ) A-law/U-law, G.722 (48K), G.729 Annex A – CS-ACELP
- Voice on LAN: SbRTP[1]/DMVS[2]: 7KHZ enhanced voice bandwidth or 3KHZ low bandwidth
- Typical latency over LAN 6ms (using 1ms packet sizes)
- Voice on WAN: Unicast network supporting UDP
- Max Packet Loss on the LAN 5%
- Bandwidth optimisation techniques: VAD (Voice Activity Detection)
- Diffserv (RFC 2474) –Type of service field configurable

Voice Characteristics
- Voice Frequency Range 50Hz – 7KHZ
- Microphone Sensitivity 63 ± 3dB
- Loudspeaker Output 2W RMS ~73dBA @.7m

Input Devices
Standard
- Built-in open microphone
- Built-in loudspeaker

Optional
- Speakerbus pluggable gooseneck microphone
  - Close Talking Noise Cancelling Microphone (52-09-021) 550mm, 8.5mm (21.6 x 0.33 in) Diameter
- Speakerbus noise cancelling handset (SE HSETM-D)
- Third party handset/Headset Support

System Management
- Managed using either:
  - iManager CMS or
  - Cisco CUCM PBX
- SNMP status monitoring

Housing
- Black ABS/Steel

Dimensions
- Width: 160mm
- Height: 200mm
- Depth: 50mm
- Weight: 1.2kg

Power Requirements
- PoE Class III IEEE 802.3af
- Typical: 6.49W
- Maximum: 15.4W

Interfaces
- Network interface 10/100BASE-T Ethernet auto sensing with PoE (RJ45 Socket)
- Handset interface port (RJ11 Socket)

Environmental

Rotary volume channel control, navigation and mode keys.


Illuminating LED indicators, home key & high resolution colour screen.

Alpha numeric keypad & fixed function keys.

- Operating temperature 5°C – 35°C (41°F 95°F)
- Relative humidity 10% - 85% RH, non condensing
- Storage environment Temperature 0°C - (32°F - 140°F), Humidity 10% - 85 % RH

Declaration of Conformity
- EN60950-1, EN55022 and EN55024.

1. iCS and Cisco registrations currently have limited functionality. Full functionality when implemented become a licensed feature in a future release.
2. Only calls placed on-hold on the SE 708 can be retrieved.
3. Speakerbus Real Time Protocol
4. Dynamic Multicast Voice Services

iD808 Intercom Specifications

Channel Capacity
- 10 Simultaneous calls (Telephony, Private Wires (ARD / MRD) and Hoot 'n' Holler)
- 8 Speaker channels (All call types)
- 11 Party conference (no additional conference equipment required)

Call Types
- Telephony
- Intercom
- Hoot 'n' Holler
- Private Wires - ARD (Automatic Ring Down) and MRD (Manual Ring Down)

User Features
- Speakerbus iManager Appliances[1]
- Avaya Aura Session Manager intergration[1]
- Cisco Unified Communications Manager[1]
- Integrated Intercom (Point 2 Point, Group Calls)
- Intelligent pagination
- 100 pages, 64 entries per page
- Max 600 configurable keys per iTurret
- 29 dedicated fixed function/hard keys
- 24 soft keys (including up to 8 x Speaker channels)
- 8 independent channel volume controls; 1 per speaker channel
- Master volume control
- Support for barge-in and privacy on private wires and telephony
- Support for programmable paginating and non paginating keys
- Style based visual architecture
- 32 Alert profiles
- Multiple speaker muting options
- Private directory with up to 1000 entries
- Corporate directory with up to 15000 entries
- Group directory with up to 2500 entries
- LDAP integration with Active Directory
- Line labelling up to 40 characters
- Single and double line styling
- 16 programmable colour styles
- Inbound caller ID matching and display
- Playback - speaker playback between 5-30 seconds of received audio[2]

Recording
- 7 Configurable IP recording streams (per iD808)
- 7 Resiliency IP recording streams (per iD808)
- Call Data Records (CDR) output (per iD808)[3]

Network
- IP Addressing: dynamic or static
- Other supported network protocols: Ethernet, IPv4, DHCP, TCP/IP, DNS, 802.1p, 802.1q and SNMP

VoIP Media
- SIP (using standards based RFC's)
- RTP supported codecs: G.711 PCM (3.4KHZ) A-law/U-law, G.722 (48K), G.729 Annex A – CS-ACELP
- Voice on LAN: SbRTP[4]/DMVS[5]: 7KHZ enhanced voice bandwidth or 3KHZ low bandwidth
- Typical latency over LAN 6ms (using 1ms packet sizes)
- Voice WAN: Unicast network supporting TCP/UDP
- Max Packet Loss on the LAN 5%
- Bandwidth optimisation techniques: VAD (Voice Activity Detection)
- Diffserv (RFC 2474) –Type of service field configurable

System Management
- Web based centralised management application
- Embedded user configurable menus
- SNMP status monitoring
- Upgradeable operating firmware

Voice Characteristics
- Voice Frequency Range 50Hz - 7KHZ
- Microphone Sensitivity 63 $\pm$ or 3dB
- Loudspeaker Output 2W RMS

Input and Output Devices
Standard
- Built-in open microphone
- Speaker with up to 8 mixed channels

Optional
- Speakerbus pluggable gooseneck microphone
  - Close Talking Noise Cancelling Microphone (52-09-021) 550mm, 8.5mm (21.6 x 0.33 in) Diameter
- Speakerbus noise cancelling handset (SE HSETD-C)
- Third party handset/wired or wireless headset Support[6]

Housing
- Desktop mounted with adjustable height from 25 - 60 degrees
- ABS plastic

Dimensions
- Width: 340mm
- Height: 200mm
- Depth: 50mm
- Weight: 2kg

Power Requirements
- Voltage 90 - 264VAC nominal
- Frequency 47 - 63Hz AC
- DC Output 12 v 2.5A (30w)

Interfaces
- 2 Switched network interface 10/100Base-T Ethernet auto sensing ports (RJ45 Sockets)
- 2 Handset interface ports (RJ11 Sockets)
- 8 Pin Mini Din Com Port (reserved for use by Speakerbus)
- 2.5mm DC outlet socket

Environmental
- Operating temperature 0°C - 35°C (32°F - 95°F)
- Relative humidity 10% - 90%, non condensing
- Restriction Of Hazardous Substances (ROHS) directive compatible

1 Please contact Speakerbus for a comprehensive list of features
2 Playback duration configurable in iManager Centralised Management Server (iCMS)
3 Requires iManager Call Detailed Service (iCDS)
4 Speakerbus Real Time Protocol
5 Dynamic Multicast Voice Services
6 Certified for use with selective Plantronics and GN Netcom headsets

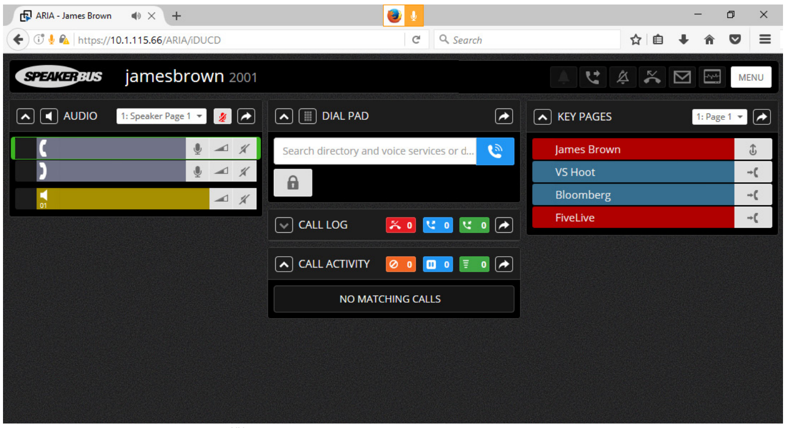
ARIA Soft Client Specifications

End User Features

- Call Types:
  - Telephony
  - Intercom
  - Group Calls
  - Hoot 'n' Holler
  - Private Wires - ARD (Automatic Ring Down) and MRD (Manual Ring Down)
- Browser based HTML5 compliant application featuring iSeries SLATE^{UX} interface
- Supports multiple audio devices
- One click show/hide individual panels
- Dock/undock individual panels
- Seated user and device status information
- Advanced logging
- Incoming call notification
- Persistent user preferences between iD808 and web browser interface
- Browser closure notification
- Customisable user interface
- Corporate or personal telephony directory access
- Dial pad for telephony calls
- Combined corporate/personal/voice service search
- Choice of host device:
  - Remote use of iD808 Trading Turret or
  - iManager Deskstation Unified Communications Virtual Endpoint (iDUCX)

Handset

- Two virtual handsets
- Visible indication of selected handset
- Latched microphone muting
- Individual volume controls
- Visible call status for all call types
- Connect an audio feed, hoot or ringdown from a list of available voice services
- Audio and microphone activity indicators
- Make a telephony call from the corporate or user's personal directory
- Make a telephony or virtual private wires (VPW) call from a speed dial
- Place a telephony or virtual private wires (VPW) call on hold
- Make a telephony or virtual private wires (VPW) call private
- Seize a line to a handset by manually entering a dial number
- Dial a number from the keyboard or virtual dial pad
- Auto select idle handset on answering a call
- Dedicated clear keys
- Transfer telephony call



ARIA's SLATE^{UX} web user interface: Speakerbus' universal user experience

- MRD signal key
- Signal DTMF tones on an active telephony call
- Handset keys dimmed whilst connecting

Speaker

- Upto 24 virtual speakers
- Individual speaker latched muting and volume controls
- Call status indication:
  - Busy elsewhere
  - Private
  - Connecting
  - Busy idle
  - Conference
- Audio and microphone activity Indicators
- Remove appearance
- VPW/telephony/intercom/group call audio feed:
  - Listen/talk
  - Move to selected handset
- Connect an ARD
- Line seizure onto selected handset to make telephony call
- Answer alerting (ringing) call
- Barge into busy elsewhere call
- Clear calls
- Talk ready indication
- One click global mute/unmute microphone for all speakers
- Page selection indicator
- Show/hide all speaker channels

Audio

- Single configurable microphone input
- Single audio output

Call Log

- View placed, received and/or missed calls
- Filtering of placed, received and/or missed calls
- Simple or detailed call log view
- Delete individual or all log entries

Call Activity

- Dynamically populated view of call and audio feed activity
- Answer alerting (ringing) call on handset
- Barge into busy elsewhere call on handset
- Take call off hold and put on handset

Key Pages

- View pre-defined or combination of line/speed dial/VPW/voice services keys
- Key page selection to choose a different set of keys
- View keys configured to appear persistently irrelevant of key page selection (fixed keys)
- Seize telephony line associated with a telephony key onto the selected handset
- Use a speed dial to make a telephony call on the selected handset
- Visible indication of which audio device an appearance is associated with (if any)
- Connect an audio feed, hoot or ringdown from a key onto the selected handset
- Visible call status for all call types on a key page key which have a relevant telephony appearance, VPW appearance or voice service appearance associated with them
- Use a VPW "speed dial" to make a VPW call on the selected handset

ARIA Soft Client Specifications

- Immediate call transfer using dedicated speed dial keys
- Answer an alerting (ringing) call on a handset using associated appearance key
- Barge into a busy elsewhere call on a handset using associated appearance key
- Take call off hold and put on handset using the associated appearance key

Interactive Indicators
- Temporary alert muting
- Call forwarding (always, no answer and on busy)
- Do not disturb
- Missed calls
- Voice mail
- Session controller status

Security and Conformance
- Encrypted logon via iCMS' own embedded secure logon
- Compliance - maintains existing Voice Recording activity using Speakerbus recording and analytics interface

Prerequisites
- Session controller, either:
    - iTurret (iD808) per user[1] or
    - iManager Device Unified Communication Session controllers (iDUCX)
- iManager Communications Server (iCS)[2]
- iManager Centralised Management Server (iCMS)[3]
- iG330/334 Media Gateway - for private wires
- Client browser[4]

Client Browser Requirements
- WebRTC and JavaScript™ enabled browser[3]
- Minimum screen resolution of 1376x768 (1920x1080 is highly recommended)
- Speaker/Microphone or Headset

iManager Web Server (iWS) Features
- Enables deployment of ARIA Web Interface to web browsers via HTTPS

iManager Gateway Server (iGS) Features
- Linux® based WebRTC (Web Real-Time Communication)
- Enables iD808 Trading Turret to be used through the ARIA web interface
- STUN and TURN support

iWS Minimum Requirements
- Processor: Intel® Xeon® (E5520 8Mb Cache 2.26Ghz)
- Minimum 2GB RAM
- Minimum disk 100GB
- SVGA 1024 x 768 display
- Operating System:
    - Microsoft® Windows Server® 2008 R2 64 bit Standard/Enterprise
    - Microsoft® Windows Server® 2012 R2 64 bit Standard/Enterprise
- IIS Internet Information Services

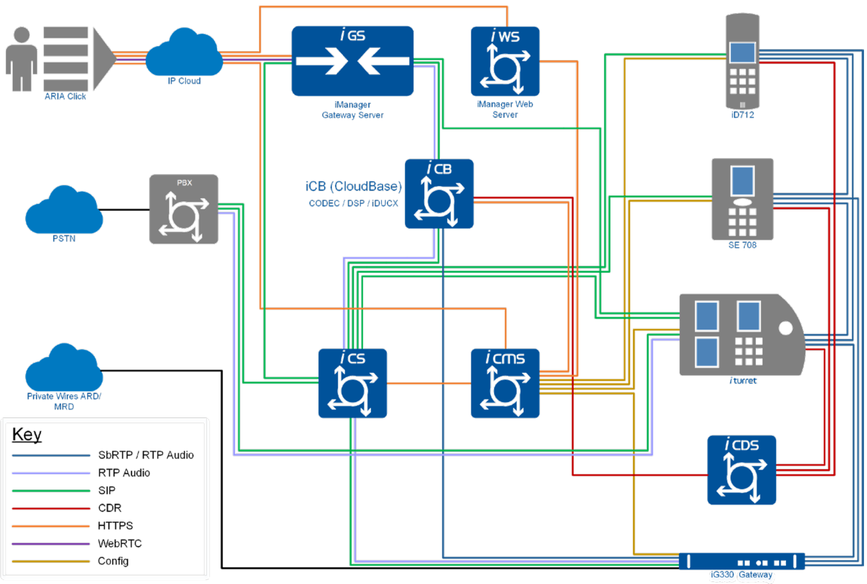
iGS Minimum Requirements (for 350 connected users)
- Processor: Intel® Xeon® (E5520 8Mb Cache 2.26Ghz)
- Minimum 2GB RAM
- Minimum disk 100GB
- SVGA 1024 x 768 display
- Operating System:
    - Linux® CentOS®
- 100Mb Ethernet (1Gb is recommended)

iManager CloudBase (iCB) Features
- Provides the services necessary for call control and media handling required to host virtualised endpoints within ARIA.
- Consists of:
    - Server management
    - Media servers
    - iSeries Device Unified Communication Session Controllers (iDUCX)

iCB Minimum Requirements (for 10 concurrent ARIA sessions)[5]
- Processor: Intel® Xeon® (E5520 8Mb Cache 2.26Ghz)
- Minimum 4GB RAM
- Minimum disk 100GB
- SVGA 1024 x 768 display
- Operating System:
    - Linux® CentOS®
- 1Gb Ethernet

Bandwidth (from browser to ARIA iGS/iWS)
- Bandwidth per stream
    - G.711/G.722 20ms 80 kbits/s
- Bandwidth upto three streams
    - G.711/G.722 20ms 240 kbits/s

1. iD808 firmware v3.000.9+ required
2. iCS software v2.4+ required
3. iCMS software v3.400.8+ required
4. Tested and supported on Windows® devices using Google Chrome® v54 and Mozilla Firefox® v50® devices using Google Chrome® v54 and Mozilla Firefox® v50
5. Alternatively iD808 iTurrets can be used instead of CloudBase (iCB/iDUCX)



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