FACTSHEET

SIPxtream™

Hardened Voice and Video Communications

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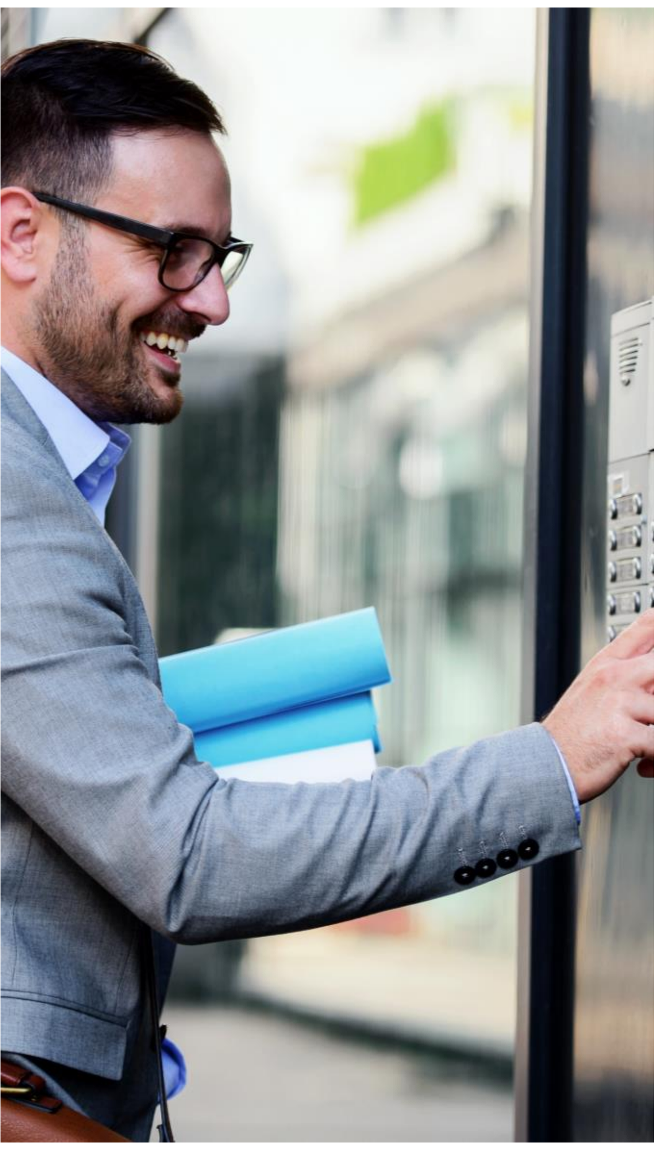
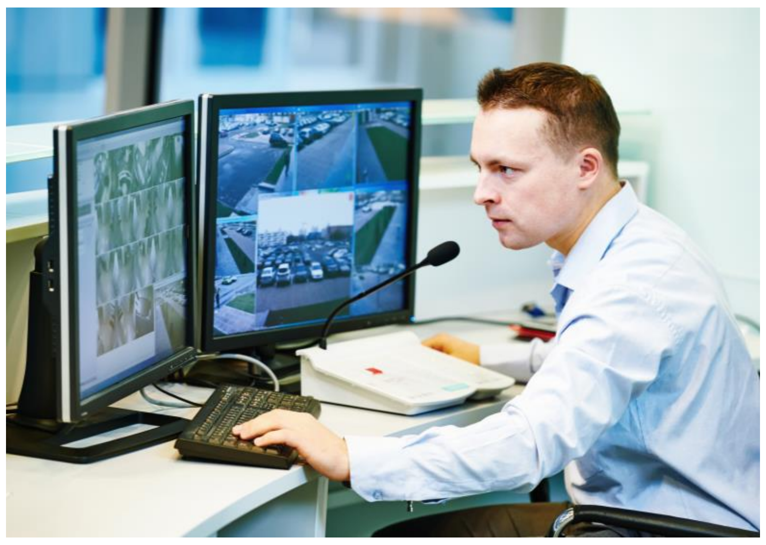




OVERVIEW

SIPxtream is communications software for mission critical voice and video devices. It offers a full-suite of endpoint services including media streaming, encoding algorithms, signalling and security. It supports audio intelligibility enhancements as well as features to provide hardened operation.

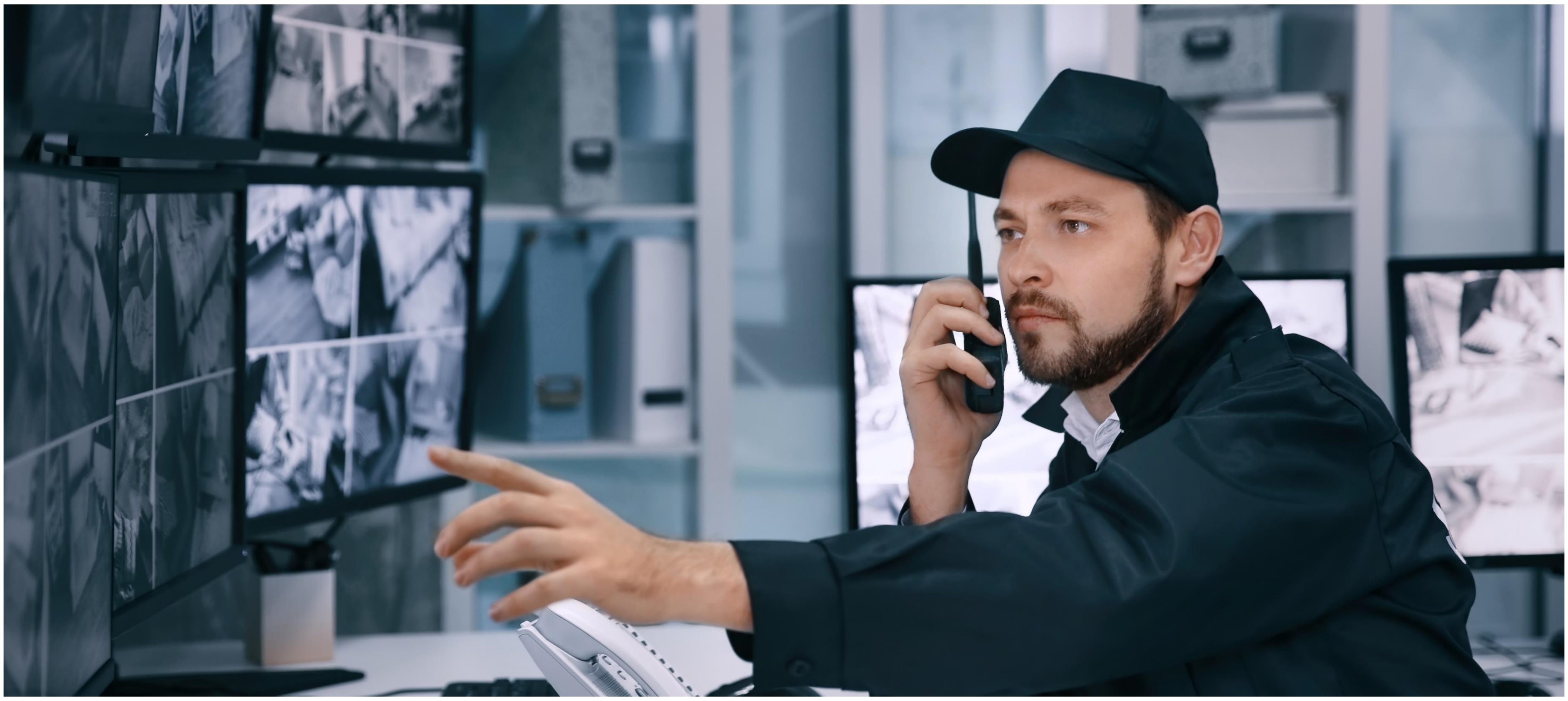
At its core, SIPxtream supports services for standards-based VoIP, multicast PA paging, recorded announcements and video. Services operate concurrently (multiservice) or multiple instances can support multichannel. The software is modular to accommodate a range of hardware from microcontrollers to AArch64 Linux systems.

SIPxtream's VoIP service offers full-duplex endpoint compatibility including SIP signalling and call flows for P2P and SIP server (PBX) interoperability. SIPxtream audio processing contains ITU codecs along with algorithms to improve audio intelligibility in applications such as intercoms. Security extensions provide fully encrypted sessions, while UVC or MIPI camera support provides video integration along with video streaming services.

Two multicast PA services are available; a receiver-only solution (compatible with standard PBX "paging"), or a hardened protocol-based transmitter/receiver for stand-alone operation and PTT.

To provide additional hardening, the system includes self-test modes, call quality detection, reporting, redundancy, failover and degraded operating modes.

- ✔ TELEPHONY, INTERCOM, PA, PTT
- ✔ VOIP, AUDIO, VIDEO STREAMING
- ✔ SIP, WEBRTC, MULTICAST SERVICES
- ✔ CODECS AND ALGORITHMS
- ✔ SECURE COMMUNICATIONS
- ✔ HARDENED FUNCTIONALITY
- ✔ RTOS AND LINUX SUPPORT



SERVICES

VoIP

Standards-based VoIP endpoint service using SIP signalling and RTP media to support full-duplex audio communication. The VoIP service consists of:

- Signalling and media protocols
- Call handling and calling features
- Peer-to-peer (P2P) or PBX server operation
- G.711a, G.711u, G.722 (wide band) codecs
- Optional G.729AB, iLBC, G.726-32, G721, AMR-NB/WB, OPUS codecs
- SIP-TLS and sRTP security with SIPS extensions
- SIP server redundancy or fallback
- Call quality information

Multicast PA – Receiver Only

PBX compatible multicast audio receiver service:

- Up to 10 groups
- Priority hierarchy from 1 to 10
- G.711u audio codec
- Compatible with standard PBX “paging” services

Multicast PA – Transmitter / Receiver

Half-duplex, multicast transmitter and receiver service. Implements a multicast control protocol for standalone operation. Suitable for multi-group mass notification, PA or group PTT applications.

- Up to 100 groups
- Last-in wins receiver arbitration
- G.711u , G.722 (wide band) audio codecs

Recorded Announcements

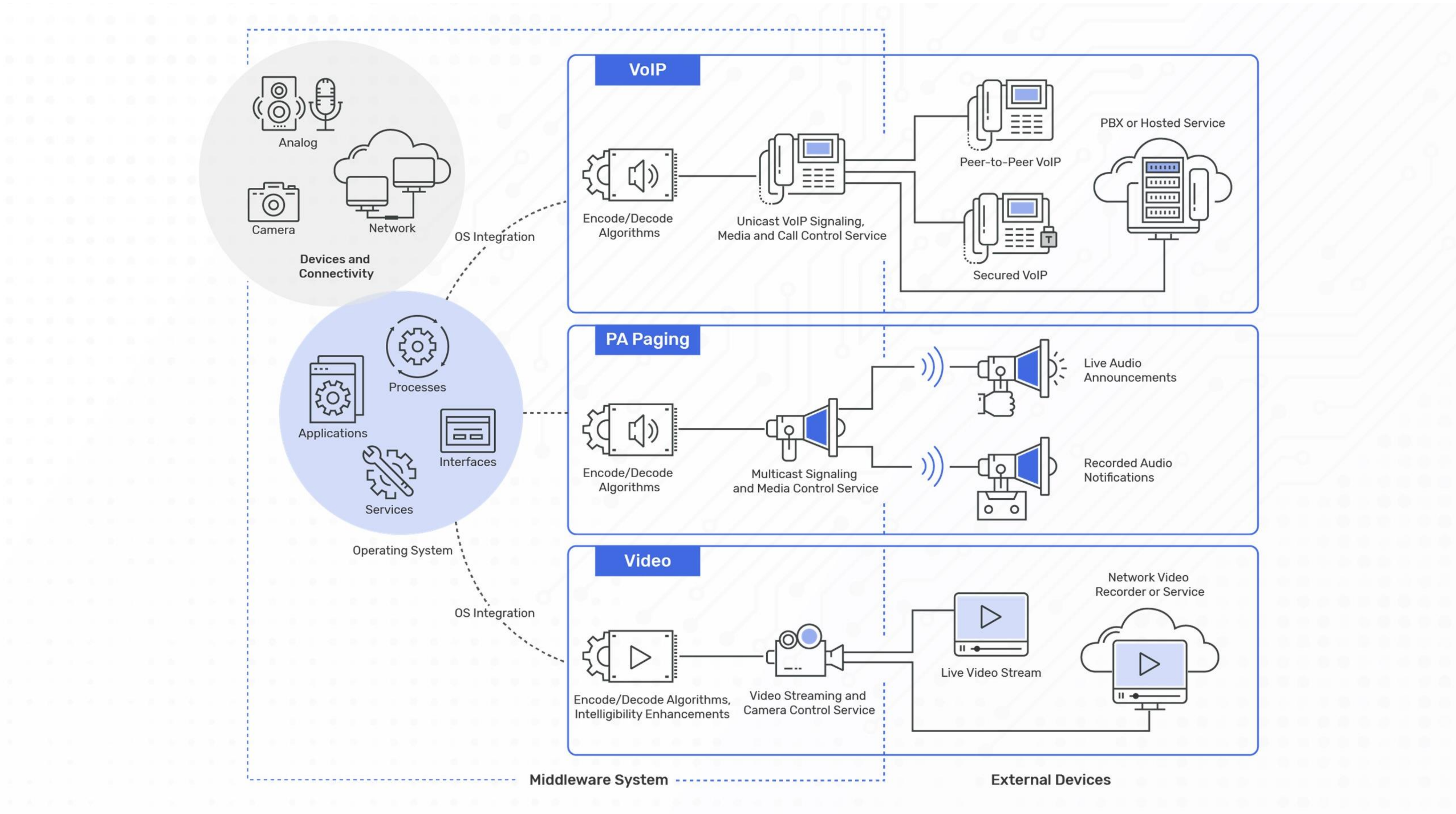
Recorded announcements provide a method to playback audio files on-demand or by controlling rate and rotation.

- Audio file playback (G.711u encoded)
- Trigger or scripted playback methods
- Local device filesystem (with management tools)
- Playback to analog, RTP or both

Video

Video support includes camera input and multiple streaming service options:

- UVC or MIPI camera integration
- Video encoding using H.264 or mJPEG
- VOIP (SIP) for video calls
- WebRTC for web browser compatibility
- RTSP streaming
- mJPEG streamer with frame rate scaling



SYSTEM INTEGRATION

SIPxtream uses web or IoT methods for configuration and management. It is also compatible with the Mbarx™ ecosystem of tools to simplify local or remote management. The Mbarx host protocol (ASD) is used for integration with a system level application or via connectivity over TCP/IP (TLS) socket or UART connection. Dedicated I/O is available for stateless operation to simplify devices that do not require host control. Linux systems offer a dedicated application partition for deeper system integration.

Integration Options

- I/O (stateless push buttons and LEDs)
- Auto-answer / hang up
- UART - host protocol (Mbarx ASD)
- TCP/IP socket (TLS) – host protocol (Mbarx ASD)
- Host application integration (Linux)

Configuration Options

- WebUI (Linux)
- Host protocol (Mbarx ASD)
- Mbarx System Manager tool

Hardening Features

- Active call quality monitoring
- Analog loop-back test for audio component performance
- SIP server redundancy
- SIP server failover
- P2P + server hybrid coexistence

Mbarx System Manager Tool

- Secure site-wide device management
- Local and remote management
- Active monitoring
- In-field firmware updates
- Windows® and Mac® compatible

VOIP FEATURES

Calling Features[1]

- PBX or P2P compatibility
- Server registration / authentication
- Digit map or direct SIP URI calling
- Call origination and termination
- Caller-ID name, date, time
- Caller-ID privacy
- Caller-ID block
- Anonymous Call Rejection (ACR)
- Message Waiting Indicator (MWI)
- Call hold and retrieve (with held call ring back)
- Call Waiting (CW)
- Cancel Call Waiting (CCW)
- Call transfer – unattended
- Call transfer – attended (REFER)
- Call transfer – attended (REPLACES)
- Failed transfer ring back (NOTIFY)
- Do Not Disturb (DND)
- Auto Call-Back on busy (ACB)
- Call Return
- Local conference bridging (3WC)
- Remote conferencing (meetME)
- Phonebook
- Speed dials
- Auto-answer
- Auto-hangup
- Hotline (hook switch input)
- PTC (button press inputs)
- Outgoing call blocking rules

SIP Compatibility[1]

- SIP, SDP, RTP, RTCP
- SIP via UDP or TCP
- VoIP Security (SIP-TLS, SIPS, sRTP)
- STUN, TURN, ICE firewall traversal
- DTMF detect and generation analog, in-band, RFC2833 or INFO methods
- Standard SIP timers
- SIP methods: INVITE, ACK, BYE, CANCEL, REGISTER, INFO, SUBSCRIBE, NOTIFY, OPTIONS, REFER, REPLACES in REFER, Reinvite Support.
- SIP text messaging
- Early media support
- Late media support

[1] some features are platform dependent

ALGORITHMS[1]

- Narrow-band (8KHz) ITU voice codecs (G.711a, G.711u)
- Wide-band (16KHz) ITU codecs (G.722)
- Optional codecs (G.729A, iLBC, G.726-32, G721-RFC3551, AMR-NB, AMR-WB, OPUS)
- Silence suppression
- Comfort Noise Generation (CNG)
- Discontinuous Transmission (DTX)
- Active Talker Detection
- Acoustic Echo Canceller (AEC)
- Noise Reduction (NR)
- Auto Gain Control (AGC)
- Dynamic Range Compression (DRC)
- Background audio level detection
- Analog Loop Back Test (ALBT) analog component performance
- Configurable codec preference order (VoIP)
- Call progress tones - dial tone, trying, ringback, busy, howler, stutter
- Peak audio level detection
- Digital (PCM) audio mixer with virtual channels

SYSTEM SOLUTIONS

For specialized solutions Arcturus provides access to our in-house communications experts, application architects and hardware designers. Engagements provide a way to access a diverse set of pre-existing technologies and expertise in enablement, algorithm creation, optimizations or complete system solutions. This approach helps manufacturers reduce project risk and decrease time-to-market, translating into a lower total cost of ownership. Arcturus offers simple engagement packages to help get development moving quickly.

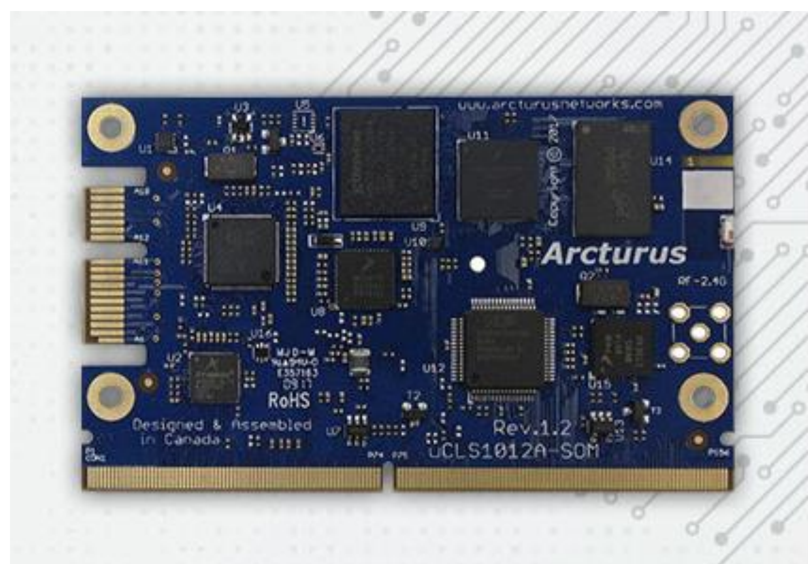
ARCTURUS HARDWARE

Whether you are designing your own chip-down solution or leveraging our boards and modules to accelerate time-to-market, Arcturus offers hardware, services and support to help. We have 100,000's of devices in service and we leverage this expertise to help you.

- Quality - Field failure rates as low as 0.1% / 10,000 units
- Longevity - Product life-cycles in excess of 20+ years
- Time-to-Market - Boards and modules ready to integrate
- Reduced Risk - Hardware and software expertise, support
- Lower Cost of Ownership - Software ready for deployment, reduces development cost

uCMK64-VoP Module

uCMK64-VoIP module is a real-time hardware platform that uses a 32-bit K64 microcontroller from NXP. It comes preloaded with SIPxtream and Mbarx Secure IoT firmware making it ready for deployment out-of-the-box. uCMK64-VoIP supports 1 VoIP channel and uses an I/O interface or host protocol for integration and management. It is simple to use with no BSP or code development required. The development kit includes everything you need to get started.



uCLS1012A-VoIP Module

uCLS1012A-VoIP is a Linux based VoIP module that uses a low-power, 64-bit LS1012A digital networking processor from NXP. It comes preloaded with SIPxtream and Mbarx firmware making it a powerful communications platform with integrated management and telemetry, including a web UI. The uCLS1012A-VoIP supports up to 2 VoIP channels, VoIP security, advanced features and optional video. The development kit includes everything you need to get started and for deeper integration, access to a Linux BSP is available with support.

[1] some features are platform dependent

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