LINK

TRULY MOBILE AUDIO OVER IP

IT'S LIKE AN OB VAN IN YOUR POCKET

RADIOKIT

IDEA

LOW-LATENCY LIVE TRANSMISSION TOOL IN YOUR POCKET.
LINK REPLACES BROADCAST VAN WITH A LAPTOP OR SMARTPHONE. NO PHYSICAL CODECS, NO WIRES, NO SATELLITE, NO ISDN – JUST INTERNET CONNECTION.

BROADCAST

Broadcast live with only a microphone plugged into your smartphone.
Be ready to interview anyone and report from anywhere at all times.

LIVE STREAM

Stream festivals, conferences and other events through multiple channels with just a laptop.

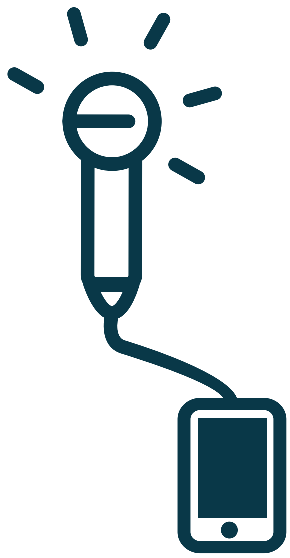
WEB CONTROL

Remotely monitor, mix, and manage multiple, collaborative broadcasts through a single web application.

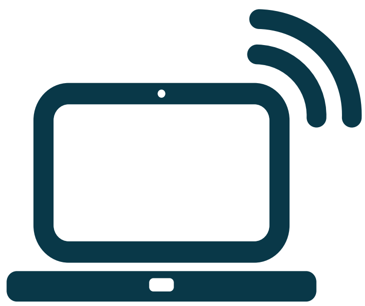
ROUTING


Event
Reporter 1
Reporter 2
LINK CLOUD
Control through web browser
Antena
Mixer

IT IS ALL A REPORTER



OR ENGINEER



NEEDS FOR GOING ON AIR

THE LATEST ACHIEVEMENTS IN AUDIO OVER IP BROUGHT TO THE BROADCASTING.

The near-to-zero latency combined with website control allows to take live broadcasting to a new era.

State-of-the art Opus 1.1 codec that brings better quality than HE-AAC+1 with ultra low-latency and packet loss concaelment.

Relies on well-tested cloud technologies and infrastructure: Amazon, Azure & Digital Ocean clouds & the biggest European datacenter: OVH.

Built upon Erlang – the language for building cellural networks.

Using Real Time Protocol and Real Time Streaming Protocol over UDP – no more unreliable TCP connections and shoutcast streams.

No hardware needed – everything is a software. Moreover: software that's easy to integrate thanks to open web & mobile-ready APIs.

User interface based on React.JS – technology created by facebook, made for the best comfort of snappy, collaborational communication.

FLEXIBILITY

The LINK is ready-to-use in minutes – you can simply log in, set the address of broadcast and go live.
Thanks to that journalist do not have to be tied to sound engineers.

However upon request we can integrate LINK with:

- custom audio routing scheme
- existing audio over ip technologies
- archiving systems
- authentication systems

USAGE

PRICING

75€

NET PRICE PER CHANNEL/MONTH

- 2 MONTHS FREE TRIAL
-10% DISCOUNT IN CASE OF ANNUAL PAYMENT UPFRONT
- “FAIR PRICING” MODEL / DISCOUNTS FOR NGOS AND DEVELOPING COUNTRIES UPON REQUEST

Cut your costs of making field transmissions in your FM/AM radio.

Your radio will not need anymore expensive trucks full of hardware or complicated equipment which confuses journalists.

Lower costs with bigger flexibility – can You expect more?

EMAIL SUPPORT IN WORKING HOURS OF CEST TIMEZONE INCLUDED
99% SLA
ADDITIONAL SUPPORT + HIGHER SLAS UPON REQUEST

TECHNICAL SPECIFICATION

Supported operating systems & architectures

Android 4.1 or newer, devices with ARM processor
Windows 7 or newer, 64-bit
Mac OS X 10.10 or newer, 64-bit 1
Ubuntu Linux 14.04 LTS, 64-bit 2

Networking

Client-server architecture
No servers required by default (servers are hosted by RadioKit in the cloud) 3
No network setup required by default
Built-in NAT traversal (UDP hole punching)
VPN access 4
IPv6 ready
Designed to operate with cable, WiFi and mobile (3G and newer) IP networks

Codecs

Opus 1.1 (RFC 6716)
Packet Loss Concealment enabled
Error Correction enabled
Bitrate from 5 to 510 kbit/s
- speech has broadcastable quality at 24 kbit/s
- music has broadcastable quality at 48 kbit/s

CBR or VBR mode
Sample rate from 8 to 48 KHz
Audio bandwidth: from 4 (narrowband) to 20 kHz (fullband)
Server-side audio processing engine internally uses 32-bit float samples at 48 KHz
Mono or stereo
Dynamically adjustable bitrate, audio bandwidth, and frame size

1 Coming soon
2 Coming soon
3 Enterprise installations on client's servers are also possible, but extra charges may apply
4 Coming soon, VPN is enabled upon client's request, extra charges may apply

TECHNICAL SPECIFICATION

Transport layer

Open standards: RTP (RFC 3550), RTSP 1.0 (RFC 2326)
Transmission happening over UDP whenever possible with TCP fallback

Latency

Default latency: 80 ms + network latency:
- Codec frame size: 20 ms
- Receiver jitter buffer: 20 ms
- Server-side audio processing engine: 20ms
- Reciver transmission buffer: 20ms
- Network latency (60 ms on average in Central Europe)

Can be adjusted in the following range:
- Codec frame size: 2.5-60 ms
- Receiver jitter buffer: 5-1000 ms
- Receiver transmission buffer: 5-1000 ms
- Network latency: down to +/- 20 ms 5

Monitoring & management

100% web-based monitoring panel
Support multiple users operating collaboratively
Web-based peakmeter showing reference signal from the server
Ability to adjust connection parameters via web browser during transmission

Authentication

Based on OAuth2 standard
Built-in web-based authentication system

5 May require setting up an exclusive servers for the client in the physically closer location, extra charges may apply

TECHNICAL SPECIFICATION

Routing

Multiple clients
Bi-directional audio communication with clients
Talkback

Extras

Built-in alternative communication channel: text chat Integration
All functions of the application are available through REST or WebSocket API
Open specification

CONTACT

INFOLINK@RADIOKIT.ORG
+44 XXX XXX XXX

APPS.RADIOKIT.ORG/LINK
FACEBOOK.COM/RADIOKITENGINE
TWITTER.COM/RADIOKITENGINE