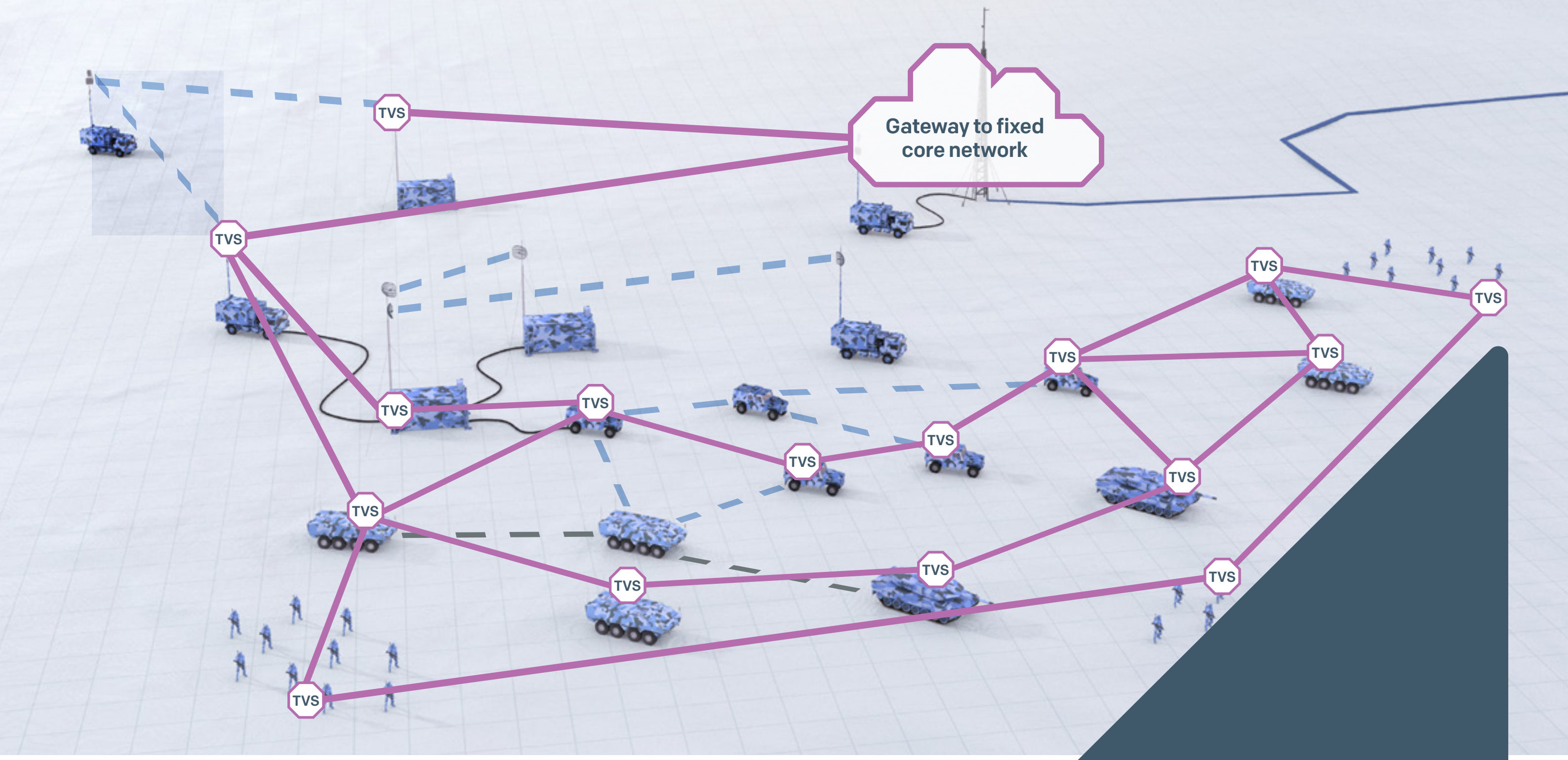
Bittium Tough VoIP Service™

Distributed VoIP Service for Tactical Networks



TVS - Bittium Tough VoIP Service node

Bittium Tough VoIP Service™ creates a highly resilient, distributed digital voice service in a tactical network. Tough VoIP Service can rapidly adapt to changes in the network. Network can be split or merged while maintaining the service available for all clients without configuration changes. When network islands merge with the network, the clients will have core network connection automatically.

Tough VoIP Service is targeted for tactical level users and the call functions have been optimized for tactical networks. Call functions include for example distributed group calls, PTT calls, call prioritization (MLPP) and broadcast calls. Provided affiliation and registration service allows user mobility and affiliation of any phone number with any client without network administration.

Benefits

- Native support for MANET
- Zero configuration
- Seamless connectivity across tactical and fixed networks
- Easy integration to other SIP based VoIP services and phones

Use case examples

- Tactical VoIP network in the battlefield
- Bridges different network technologies and vendor networks
- Voice communication to CNR network (RoIP)
- Presence status with Tough VoIP Softphone and Tough Comnode

FOR MORE INFORMATION, PLEASE CONTACT:
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Bittium Tough VoIP Service™
Specifications

Features

SUPPORTED VOIP CALLS

- Calls within multiple Tough VoIP Service nodes
- Calls between two different Tough VoIP Service networks
- Calls to/from any other VoIP systems
- Voice to/from CNR network (RoIP)

NETWORK TOPOLOGIES

- Support for arbitrary network topologies
- Adjusts automatically to network split and merge
- Support for call routing between TVS and external VoIP networks
- Support for Call Admission Control (CAC)
- Support for Quality of Service (QoS)

SUPPORTED CLIENTS

- **Bittium Tough VoIP™ products**
- COTS VoIP phones and soft-phones (for example Cisco, Linksys, Linphone and Mitel SIP phone)

SUPPORTED CALL MANAGERS

(connectivity to external VoIP networks)

- Cisco Unified Communications Manager (CUCM)
- FreeSWITCH

SIGNALING PROTOCOLS

- SIP (RFC 3261) and SDP (RFC 2327 / 4566)

VOICE AND DATA

- RTP (RFC 3550), RTCP (RFC 36551) and Telephone events (RFC 2833)
- SIP message routing (RFC 3428)
- Presence event package (RFC 3856)
- TLS

AUDIO CODECS

- G.711 A-Law, G.729, G.729A, G.729B, G.729AB, GSM, Speex, MELPe 2400

PHONEBOOK

- LDAP (RFC 451)

MANAGEMENT INTERFACE

- WEB GUI management interface
- Configuration via XML or .json configuration file
- One configuration file for all Tough VoIP Service nodes
- Representational State Transfer over https (REST-API)

SYNCHRONIZATION METHOD

- Between different multicast areas with unicast gateways
- Automatic multicast based discovery / synchronization

USER AND SERVICE MOBILITY

- Supports phone mobility between Tough VoIP Service nodes
- Supports user mobility between different VoIP phones (Affiliation and de-affiliation)
- Supports user to transfer active call to other phone

CALL FEATURES

- Affiliation and de-affiliation (hot-desking)
- User defined and predefined group calls with audio mixing or PTT controlled
- Broadcast and multichannel call
- Priority point-to-point and group calls with call pre-emption (5 classes)
- Compressed dialing
- Call forwarding, transfer and waiting
- Call hold, switch and resume
- Callback
- Hunting group
- Emergency call
- Dynamic user settings
- Traffic matrix
- CLIR/CLIP
- User-definable re-routing
- Multiple devices with the same number
- Conference rooms
- Phonebook (LDAP)

COLLABORATION FEATURES

- Meeting rooms with chat and audio
- Managed group call service with call activity detection and automatic recall

CNR FEATURES

- Support for Combat Net Radio Gateway

SUPPORTED NETWORK SIZE AND USERS

- Up to 1000 Tough VoIP Service nodes in the network
- Up to 200 VoIP users in one Tough VoIP Service node
- Up to 10 000 VoIP users in the Tough VoIP Service network

SUPPORTED PLATFORMS

- Computer with Linux OS
- **Bittium TAC WIN Tactical Router™**
- **Bittium Tough Comnode™**
- **Bittium Tough SDR Vehicular™**
- **Bittium Tough SDR Handheld™**

Product code: 9530282

Performance on **Bittium TAC WIN Tactical Router™, without transcoding**

SIMULTANEOUSLY, PER NODE

- > 50 active call sessions
- > 200 registered subscribers

Recommended Minimum PC configuration

OS

- Installation packages for Debian
- Portability to other Linux versions

RAM (RESERVED BY TVS)

- 64 MB (reserved) 64-bit build
- 48 MB (reserved) 32-bit build

STORAGE

- 32 MB (includes runtime log files)

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