Voice Play-Record for OTF Kernel

OTF Kernel is licensed without media-processing resources. PowerVox for OTF Kernel adds voice play and record capability to an Open Telecommunications Framework® system. The PowerVox OTF Kernel SDK provides the necessary development environment; runtime licenses add a specific number of per-call resources. These runtime resources can be for any MSP-series resource hardware or the terminating version of any voice coder that supports voice-over-packet networks.

PowerVox for OTF Kernel provides the OTF developer an ECTF S.100-conforming Player-Recorder API, allowing the client application to access PowerVox's functionality through an open-standard API. The resulting application is portable across the available S.100 server implementations.

The play-record resource used for a given command is determined by the OTF Kernel system on a per-call basis without involvement by the application. The resource, which is determined by the OTF resource group configured to support the operation, may be terminating IP voice using host-signal processing, the MSP-H8, or MSP-320. This means the application can be developed without regard to the type of network (PSTN or IP) or resource to be used on a given call.

OTF Kernel is an ECTF S.100 R2-conforming telephony middleware product that supports both third-party and Commetrex-provided media and switching resources. These vendor-specific resources are isolated from the balance of the OTF Kernel system by Resource Service Managers (RSMs). The resource-specific RSM determines the number of concurrent media-technology resources that have been licensed. It then launches the appropriate Resource Controllers, such as the PowerVox Resource Controller.

Features

- Field proven voice technology
- Resource- and network-independent client API
- S.100 Client API
- G.711, G.726 (16, 24, 32, 40-K bit rates, μ-law and A-law)
- Wave file support
- 8-bit linear audio (11, 22, 44-K bit)
- 16-bit linear audio (8, 11, 22, 44-K bit)
- S.100 commands (play, stop, pause, resume, jump)
- Maximum-Concurrent Port Licensing
- Software-password license administration
- Optional G.723.1 & G.729a/b

Benefits

- Application portability
- Reduced development time and cost
- High customer satisfaction
- Deployment flexibility

Overview

An OTF system provides clear separation between the application and service entities and network-interface and the resources that connect and process the digital-media call streams. These resources are abstracted to render them vendor and resource independent. The binding of the resource-abstraction token to an application and call stream is then handled by the system services independently of the application.

The OTF Kernel uses an application profile and System Call Router routing rules to dynamically bind resources to the application on a per-call basis. For example, the application profile may specify a particular voice coder to use, where routing rules allow the SCR of build a resource group for the application that would be based on whether a call arrived via a PSTN or packet connection.

Player-Recorder API

The OTF Player-Recorder provides the functions needed to implement messaging and voice-based digital-media information service platforms. The API functions operate on S.100-compatible container Time-Varying Media (TVM) objects.

CTplyr_Adjust Volume()	Adjust the current playback volume.
CTplyr_Jump()	Jump back or forward within the TVM playback.
CTplyr_Pause()	Pause playback of a TVM.
CTplyr_Play()	Start Playback from a TVM.
CTplyr_Resume()	Resume playback of a TVM.
CTplyr_Stop()	Stop playback of a TVM.
CTrcdr_Pause()	Pause recording
CTrcdr_Record()	Start Recording into a TVM.
CTrcdr_Resume()	Resume recording.
CTplyr_Stop()	Stop recording.

Coders Supported

The standard OTF speech-file is stored in the WAV file format. Supported coders are as follows:

Coder Type	Sample Size (bits)	Sample Rate
ADPCM at 16 kbps	2	8,000
ADPCM at 24 kbps	3	8,000
ADPCM at 32 kbps	4	8,000
ADPCM at 40 kbps	5	8,000
µ-law PCM at 64 kbps	8	8,000
A-law PCM at 64 kbps	8	8,000
11 kHz 8 bit linear audio	8	11,000
22 kHz 8 bit linear audio	8	2,2000
44 kHz 8 bit linear audio	8	44,000
8 kHz 16 bit linear audio	16	8,000
11 kHz 16 bit linear audio	16	11,000
22 kHz 16 bit linear audio	16	22,000
44 kHz 16 bit linear audio	16	44,000

ITU G.726 ADPCM

This ITU standard describes the conversion of a 64-Kbit A-law or µ-law PCM channel (sampling rate = 8000Hz and sample size = 8 bits) to and from a 40, 32, 24, or 16-Kbit/s channel. The conversion is applied to the PCM bit stream using the ADPCM transcoding technique described in G.726.

Uniform Pulse Code Modulation (UPCM)

In uniform PCM, each sample of the incoming signal is quantized to one of 2^R amplitude levels, where R is the number of binary digits used to represent the sample. For example, in 8-bit uniform PCM each sample is quantized to one of 256 levels.

Configuration Information

PowerVox adds terminating voice to an OTF Kernel-based system. All members of the MSP line of DSP-resource boards are supported, as is the all-IP BladeWare IP media server.

OTF for MSP SDK, PN 20070
OTF PowerCall SDK, PN 20050
OTF PowerVox SDK, PN 20060
PowerVox Runtime License, PN 50006

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