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webRTC源码简要目录

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- | |—— audio_mixer 【混音，比如音视频通信时，同时有多个人讲话，为了方便传输和管理，会把多路声音混在一起统一传输】
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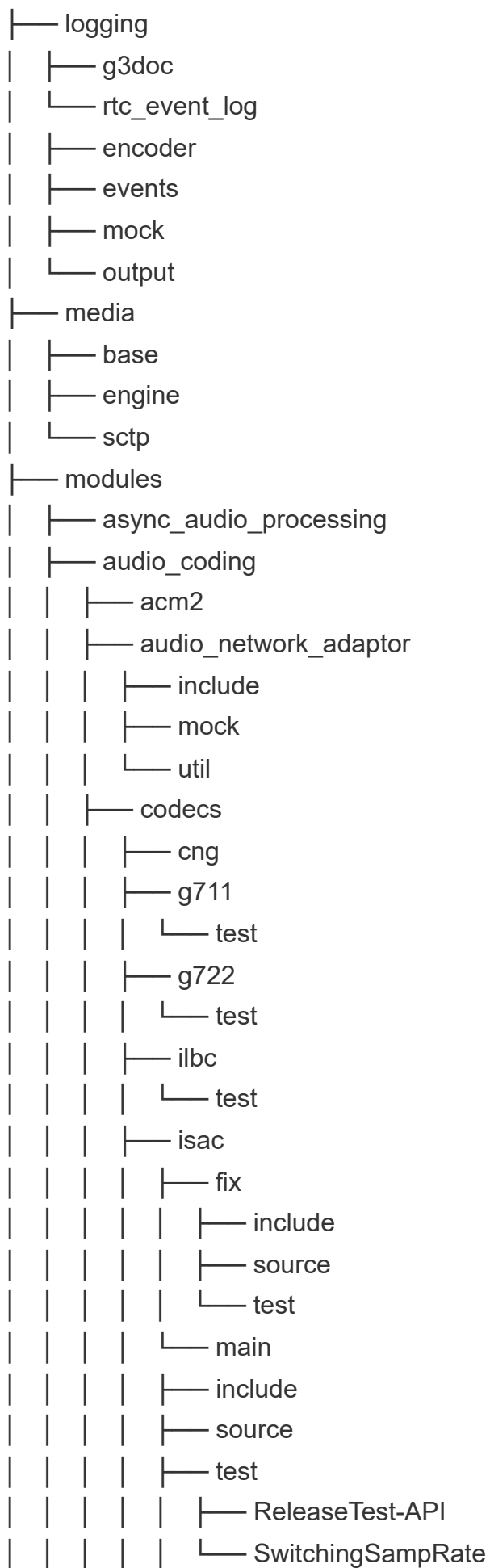
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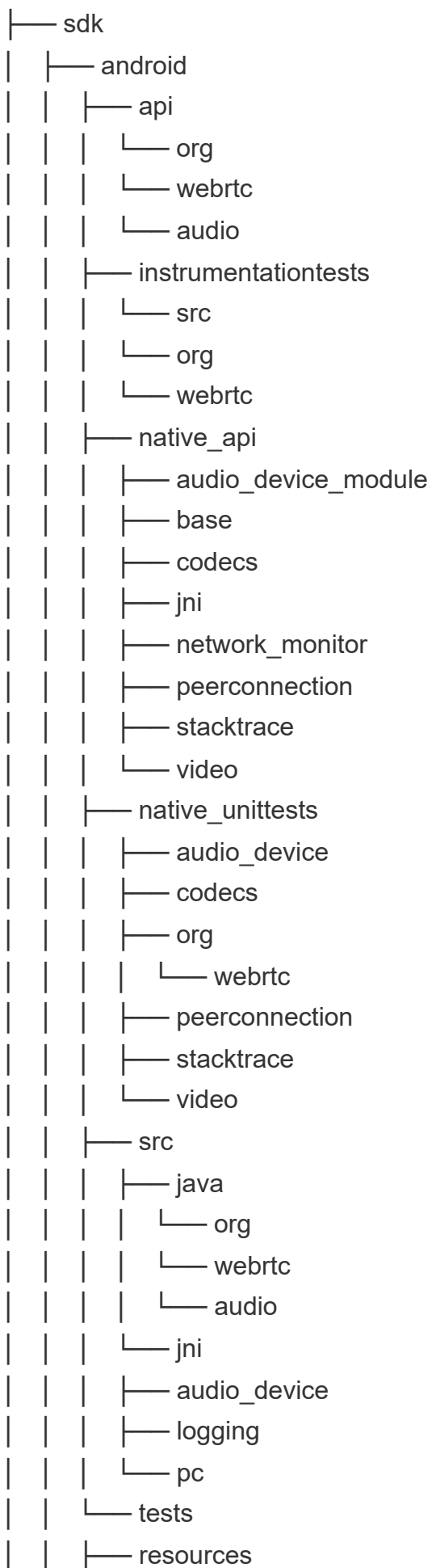
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